JVC

PRODUCT REFERENCE GUIDE

1990

HI·FI

This guide has been prepared to give you an understanding on JVC Hi-Fi products for 1990, the technology used and our new lineup, etc. It consists of the following seven sections:

1. The feature highlights of '90 JVC Hi-Fi products including sales points, with photographs and illustrations.
2. Feature comparison charts. — These show the positioning of products in relation to one another.
3. Lineup by product category, with itemized features and photographs of each product; one-point features and logos attached show the functions of each model at a glance.
4. Feature reference with simplified description of each feature/technology and its benefits; this section gives brief explanations of the technologies and features of our products.
5. Description of important technologies; this section gives more details of technologies and features that we are emphasizing this year.
6. For certain products, we include examples of how to conduct effective demonstrations.
7. At the end of this book, "New Hi-Fi Technology 1990" gives detailed descriptions of important new technologies.

We hope that this guide will give you a fuller and better appreciation of the quality of JVC's lineup of Hi-Fi Audio Products and help you in your sales activities.

CONTENTS

RECEIVERS

Feature Highlights of '90 Receivers

1 ***CSRP improved, with factory-preset values added to the comprehensive display facility; the user can see the settings and use it more easily***

2 ***Programmable remote control unit enables direct access to any A/V component; this year, our programmable remote control unit is improved with a new sequential memory function for enhanced operability***

3 ***Comprehensive A/V functions including Dolby Surround (with Pro-Logic Circuitry), a 4-channel amplifier and digital acoustics processing system give your system another dimension***

Improved CSRP (COMPU LINK Source Related Preset) function

With this, the optimum values of the settings of volume, balance, loudness, SEA, surround, etc. for each source can be stored in memory and recalled immediately while the settings will be shown on the FL display in sequence, whenever the corresponding source is selected. As factory-preset values for each source are stored in the memories of these receivers, these can be recalled when the receivers are switched on for the first time. Therefore, these values can be used as they are as well as modified to the user's preference. In addition, as the remote control units provided with most receivers incorporate a convenient CSRP display, you can check the values and perform adjustment without leaving your listening position. Also, a CSRP test (demonstration) function is provided in the RX-1010VTN and RX-903VBK so that the factory-preset and user preset values can be shown in sequence on the FL display until the function is released. This function is extremely effective in an in-store display.

Programmable remote control for total control over other JVC components and components from different manufacturers

The programmable remote control features a "learning" function which allows it to store the functions of most components from JVC and even from other manufactures. After it has "learned" the functions, they can be recalled at any time so that remote control over the entire system is possible with a single unit. Another advantage of this year's remote control unit is its sequential memory function with which 6 sequences of operations can be stored in memory.

DOLBY SURROUND PRO · LOGIC

Enhanced A/V functions make the receiver a versatile A/V control center

In addition to the basic facilities of the receiver, it has a full range of A V sophisticated control functions. For example, Dolby Surround with Pro-Logic Circuitry and 4-channel amplifier bring movie theater excitement into home listening rooms while the digital acoustics processing system adds an extra dimension for the ultimate control of acoustic ambience. It lets you enjoy movies with a more realistic sound field, closer to ever the sound you hear in a movie theater

Feature Comparison Chart

Receivers

		RX-1010VTN	RX-903VBK	RX-803VBK	RX-703VBK	RX-503BK	RX-403BK	RX-302BK	RX-201BK
Power Output									
Watts per Channel	**2-Channel Operation (Frequency Response)**	120 W (20-20kHz)	100 W (20-20kHz)	120 W (20-20kHz)	100 W (20-20kHz)	80 W (20-20kHz)	60 W (20-20kHz)	60 W (40-20kHz)	40 W (40-20kHz)
4-Channel Operation	Front	110 W	90 W	110 W	90 W				
	Rear	15 W	15 W	12 W	12 W				
THD (%)	2-CH Rated	0.007	0.007	0.007	0.007	0.03	0.03	0.03	0.08
Dynamic Power		360 W (2 ohms)	300 W (2 ohms)	220 W (4 ohms)	200 W (4 ohms)				
Remote Control									
AV Programmable		✓ (LCD)	✓	✓					
AV Unified					✓	✓	✓		
Power ON/OFF		✓	✓	✓	✓	✓	✓		
Sleep Timer		✓	✓	✓	✓	✓	✓		
Circuit									
Dynamic Super-A (Front)		✓	✓	✓	✓				
Gm Circuit (Front)		✓	✓	✓	✓	✓	✓	✓	
Digital AP		✓							
Surround Sound	**Dolby Pro-Logic**	✓	✓						
	Dolby Surround			✓	✓				
	4-Speaker Surround					✓	✓		
S-VIDEO Terminals		✓ (2)	✓ (2)						
Center-Output		✓	✓						
Front/Rear Output		✓	✓						
Display									
CCS		✓	✓	✓	✓	✓	✓	✓	✓
FL		✓	✓	✓	✓	✓	✓	✓	✓
Source									
Audio	CD	✓	✓	✓	✓	✓	✓	✓	✓
	DAT	✓	✓						
	FM	✓	✓	✓	✓	✓	✓	✓	✓
	Phono	✓	✓	✓	✓	✓	✓	✓	✓
	Tape 1	✓	✓	✓	✓	✓	✓	✓	✓
	Tape 2	✓	✓	✓	✓	✓	✓	✓	✓
Video	VCR 1	✓	✓	✓	✓				
	VCR 2	✓	✓	✓	✓				
	Video/TV	✓	✓						
Sound Selector		✓	✓						
Function									
CSRP		✓	✓	✓	✓	✓			
CSRP Display (Remote Control)		✓	✓	✓	✓	✓			
CSRP Test		✓	✓						
SEA	**Electronic**	✓	✓	✓	✓	✓			
	No. of Elements	7	7	7	7	7	5	5	
	Preset Memory	5 + 5	5 + 5	5 + 5	5 + 5	5 + 5			
	Pattern Name Memory	✓	✓	✓	✓	✓			
	SEA Dub & Rec	✓	✓	✓	✓	✓			
Tuner	**No. of Presets**	40	40	40	40	40	40	40	40
	Numeric Keypads	✓	✓	✓	✓	✓	✓	✓	✓
	Station Name Memory	40	40	40	40	20			
	Preset Scan	✓	✓	✓	✓	✓	✓	✓	✓
	Auto memory	✓	✓	✓	✓	✓	✓	✓	✓
MM/MC Cartridge Selector		✓							
Loudness		✓	✓	✓	✓	✓	✓	Auto	Auto
Speaker Systems		2	2	2	2	2	2	2	2
Flip-down Door		✓	✓	✓	✓				
COMPU LINK Remote Control Component		✓	✓	✓	✓	✓	✓		

Feature Comparison Chart

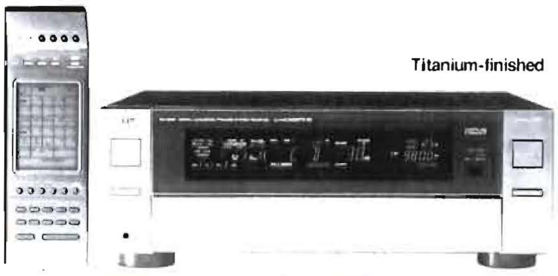
Remote Control Unit

		RM-SR1010U	RM-SR903U	RM-SR803U	RM-SR703U	RM-SR503U	RM-SR403U
General							
Provided with		RX-1010VTN	RX-903VBK	RX-803VBK	RX-703VBK	RX-503BK	RX-403BK
Programmable		✓	✓	✓			
	LCD Touch-Panel	✓					
	No. of Programs	Variable (180)	Variable (200)	Variable (163)			
	Sequential		✓	✓			
AV Remote					✓	✓	✓
Power ON/OFF (Audio/TV/VCR)		✓	✓	✓	✓	✓	✓ (Audio/VCR)
Sleep Timer		✓	✓	✓	✓	✓	✓
Volume (Up/Down)		✓	✓	✓	✓	✓	✓
Mute		✓	✓	✓	✓	✓	✓
CSRP		✓	✓	✓	✓	✓	
CSRP Display		✓	✓	✓	✓	✓	
Digital AP. Control		✓					
Surround							
Surround ON/OFF		✓	✓	✓	✓	✓	
Center Level (Up/Down)		✓	✓				
Rear Level (Up/Down)		✓	✓	✓	✓		
Mode Select		✓	✓	✓	✓		
Delay Time Control		✓	✓	✓	✓		
Source							
Audio	CD	✓	✓	✓	✓	✓	✓
	DAT	✓	✓				
	FM	✓	✓	✓	✓	✓●	✓●
	AM	✓	✓	✓	✓	✓●	✓●
	Tape 1	✓	✓	✓	✓	✓	✓
	Tape 2	✓	✓	✓	✓	✓	✓
	Phono	✓	✓	✓	✓	✓	✓
Video	VCR 1	✓	✓	✓	✓	✓	✓
	VCR 2	✓	✓	✓	✓		
	Video/TV	✓	✓				
SEA							
SEA ON/OFF		✓	✓	✓	✓	✓	
Frequency Band Adjust		7	7	7	7	7	
Level Up/Down		✓	✓	✓	✓	✓	
Tuner							
Direct Access (Numeric Keys)		✓	✓	✓	✓	✓	✓
Preset Stations Up/Down		✓	✓	✓	✓		
CD Changer							
Disc Select		✓	✓	✓	✓	✓	✓
Track Select (10-key)		✓	✓	✓	✓	✓	✓
Play Mode (CONT./PROGM/MAG. PRGM)		✓	✓	✓	✓	✓	✓
Play/Stop		✓	✓	✓	✓	✓	✓
Auto Search		✓	✓	✓	✓	✓	✓
CD Player							
Numeric Keypad (1-10, +10, 0)		✓	✓	✓	✓	✓	✓
Play/Stop		✓	✓	✓	✓	✓	✓
Auto Search		✓	✓	✓	✓	✓	✓
Manual Search		✓	✓	✓			
Pause		✓	✓	✓			
Open/Close		✓					
Turntable							
Play		✓	✓	✓	✓	✓	✓
Stop		✓	✓	✓	✓	✓	✓
Cassette Deck							
Play/Stop		✓	✓	✓	✓	✓	✓
Rec/Pause		✓	✓	✓	✓	✓	✓
FF/REW		✓	✓	✓	✓	✓	✓
Music Scan		✓	✓	✓	✓	✓	✓
Direction		✓	✓	✓	✓		
DAT							
Play/Stop		✓	✓	✓	✓		
Rec/Pause		✓	✓	✓	✓		
FF/REW		✓	✓	✓	✓		
VCR							
Play/Stop		✓	✓	✓	✓	✓	✓
Rec/Pause (Still)		✓	✓	✓	✓	✓	✓
FF/REW		✓	✓	✓	✓	✓	✓
TV							
Direct Access Channel (10-key)		✓	✓	✓	✓	✓	
Channel Up/Down		✓	✓	✓	✓		

● As the data is stored in the receiver's memory, the station listened to before the power was switched off is selected (last channel memory).

CSRP 4-channel receiver with programmable LCD remote control and Digital Acoustics Processor

Titanium-finished



RX-1010VTN

SUPER DIGIFINE COMPU LINK /// Remote /// Control Component

Remote Control Computerized Receiver

- Power output: 2-channel, 2 x 120 watts★; 4-channel, 2 x 110 watts★ (front), 2 x 15 watts (rear)☆
- Digital Acoustics Processor
- Dolby Pro-Logic surround with digital delay
- CSRP (COMPU LINK Source-Related Presetting)
- LCD AV programmable remote control

★at no more than 0.007% THD (8 ohms, 20 Hz — 20 kHz) (RMS)
☆at no more than 0.07% THD (8 ohms, 20 Hz — 20 kHz) (RMS)

NEW *CSRP 4-channel receiver with new programmable remote control and Dolby "Pro-Logic" Surround*

RX-903VBK

DIGIFINE COMPU LINK /// Remote /// Control Component

Receiver/System Control Center

- Power output: 2-channel, 2 x 100 watts★; 4-channel, 2 x 90 watts★ (front), 2 x 15 watts (rear)☆
- Dolby Pro-Logic surround with digital delay
- CSRP (COMPU LINK Source-Related Presetting)
- 3 video inputs with S-Video terminals
- New AV programmable remote control

★at no more than 0.007% THD (8 ohms, 20 Hz — 20 kHz) (RMS)
☆at no more than 0.7% THD (8 ohms, 20 Hz — 20 kHz) (RMS)

NEW *CSRP 4-channel receiver with new programmable remote control*

RX-803VBK

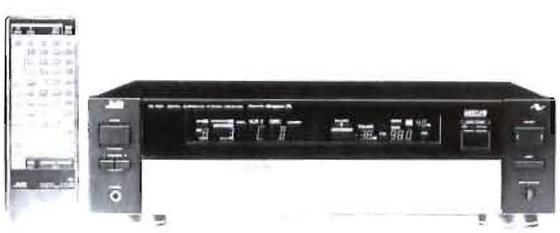
DIGIFINE COMPU LINK /// Remote /// Control Component

Receiver/System Control Center

- Power output: 2-channel, 2 x 120 watts★; 4-channel, 2 x 110 watts★ (front), 2 x 12 watts (rear)☆
- CSRP (COMPU LINK Source-Related Presetting)
- Dolby Surround with digital delay
- New AV programmable remote control
- Computer-controlled 7-band S.E.A. graphic equalizer

★at no more than 0.007% THD (8 ohms, 20 Hz — 20 kHz) (RMS)
☆at no more than 0.7% THD (8 ohms, 20 Hz — 20 kHz) (RMS)

NEW *CSRP 4-channel receiver with Dolby Surround*



RX-703VBK

DIGIFINE COMPU LINK /// Remote /// Control Component

Receiver/System Control Center

- Power output: 2-channel, 2 x 100 watts★; 4-channel, 2 x 90 watts★ (front), 2 x 12 watts (rear)☆
- CSRP (COMPU LINK Source-Related Presetting)
- Dolby Surround with digital delay
- Unified AV remote control
- Computer-controlled 7-band S.E.A. graphic equalizer

★at no more than 0.007% THD (8 ohms, 20 Hz — 20 kHz) (RMS)
☆at no more than 0.7% THD (8 ohms, 20 Hz — 20 kHz) (RMS)

NEW CSRP receiver with AV remote control and computer-controlled S.E.A. graphic equalizer

RX-503BK

Receiver/System Control Center

COMPU LINK Remote Control Component

- Power output: 2 x 80 watts*
- CSRP (COMPU LINK Source-Related Presetting)
- Computer-controlled 7-band S.E.A. graphic equalizer
- 4-speaker surround
- Unified AV remote control

*at no more than 0.03% THD (8 ohms, 20 Hz — 20 kHz) (RMS)

NEW Digital synthesizer receiver with AV remote control

RX-403BK

Receiver/System Control Center

COMPU LINK Remote Control Component

- Power output: 2 x 60 watts*
- 5-band S.E.A. graphic equalizer
- AV remote control
- Random preset memory for 40 FM/AM stations
- 4-speaker surround

*at no more than 0.03% THD (8 ohms 20 Hz — 20 kHz) (RMS)

NEW FM/AM synthesizer receiver, with S.E.A. graphic equalizer

RX-302BK

FM/AM Synthesizer Receiver

- Power output: 2 x 60 watts*
- 5-band S.E.A. graphic equalizer
- Gm driver
- FL display
- Preset memory for 40 FM/AM stations

*at no more than 0.03% THD (8 ohms, 40 Hz — 20 kHz) (RMS)

NEW Basic receiver with preset memory for 40 FM/AM stations

RX-201BK

FM/AM Synthesizer Receiver

- Power output: 2 x 40 watts*
- Four inputs — CD/PHONO/TAPE 1/TAPE 2
- Preset memory for 40 FM/AM stations
- FL display
- Connections for 2 pairs of speaker systems

*at no more than 0.08% THD (8 ohms 40 Hz — 20 kHz) (RMS)

CSRP (COMPU LINK Source Related Preset) System
(RX-1010VTN, RX-903VBK, RX-803VBK, RX-703VBK, RX-503BK)

This allows the optimum values of the settings of volume, balance, loudness, SEA, surround, etc. to be stored in memory for each input source, independently. Using CSRP, all the preset values can be recalled immediately, while the settings will be shown in sequence on the FL display, whenever the corresponding source is selected. Furthermore, since all receiver control, source selection and switching operations are done electronically, the CSRP settings can be recalled from memory using the CSRP DISPLAY function of the intelligent remote control unit provided with the receiver. Also, factory-preset values have been already stored in memory for each source and can be recalled when the receiver is switched on for the first time.

Refer to "Technical Notes" page 11 for more information.

To play different program sources in the most listenable conditions, the output level and other settings are usually different for each source. When listening to the soundtrack of a video you're watching on your TV, the settings will be different from those required when playing a compact disc. Normally, the user has to change the settings each time a new source is selected. With our CSRP system, we've solved this problem in a sophisticated way. Since all the setting values are stored in memory, independently for each input source, the optimum settings for any source can be recalled immediately. Once the user sets his or her preferred settings for a particular source, these settings are recalled by just pressing the corresponding source button. When the source starts playing, you can check the preset values by pressing the CSRP DISPLAY button; the values are displayed in sequence on the receiver's front panel. As factory-preset values for each source are provided in these receivers, they can be used as they are, or modified as required.

CSRP test function (RX-1010VTN, RX-903VBK)

A CSRP test (demonstration) function is provided in the RX-1010VTN and RX-903VBK. With this, factory-preset values or user-programmed settings for each source, such as volume, balance, SEA and surround, etc. can be shown in sequence on the FL display until this function is released.

If the CSRP test (demonstration) function is used, the factory-preset or user settings can be seen in a continuous sequence until the function is released. This function can easily be used for an effective in-store demonstration. Also, the user can check his or her settings for each source in sequence.

Programmable remote control unit (RX-1010VTN, RX-903VBK, RX-803VBK)

JVC's top-end receivers come with programmable remote controls which have a "learning" function. The remote controls provided with this year's new receivers, the RX-903VBK and RX-803VBK, incorporate a new sequential memory function which allows 6 sequences of operations to be stored in memory. With these remote control units, the user can store the functions of most components provided with infrared remote controls, even from other manufacturers, into the memory of the remote control. After "teaching" it the functions of other units, these "learned" functions can be recalled at any time, so that remote control over the entire system is possible with a single remote control unit.

As long as it uses an infrared remote control system, the functions of almost any component, even made by another manufacturer, can be memorized by our "programmable" remote controls. With this, the user can program the remote control with the functions of a number of components, and you can operate any of the components with just one remote controller. Instead of having several remote controls on your coffee table, a single JVC remote control is all you need, even if the components making up your audio/video system come from different manufacturers.

Refer to "Technical Notes" page 12 for more information.

Three video inputs including S-VIDEO terminals
(RX-1010VTN, RX-903VBK, RX-803VBK*, RX-703VBK*)

Three inputs are provided for the connection of video source components: VCR1, VCR2 and VIDEO. The RX-1010VTN and RX-903VBK are provided with S-VIDEO as well as composite video input terminals, for connection of the latest high picture-quality VCRs.

(*Two composite video inputs)

You can connect up to three video source components and switch the audio and video signals simultaneously either from the main unit or remote control, making AV integration and the dubbing of video tapes easy. As TVs and VCRs with S-VIDEO terminals can be connected to the RX-1010VTN and RX-903VBK, these two receivers are ideal for use with the latest video components.

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Digital Acoustic Processing System (RX-1010VTN)

Drawing on JVC's advanced digital technology and employing our original digital signal processing circuitry, this specially developed system allows the user to create a true, more "realistic" sound field, almost exactly the same as that obtained in a concert hall, etc., in his or her listening space. So that any of the variables intro-duced where the recording was made can be simulated, the RX-1010VTN has seven preset acoustic patterns — "Symphony Hall", "Recital Hall", "Opera House", "Church", "Live Club", "Stadium" and "Movie Theater" — while the "Room Size", "Liveness" and "Wall Type" parameters permits fine adjustment of the acoustic sound field to suit the room where the music is reproduced. With the addition of a pair of "presence" speakers, driven by the built-in rear-channel amplifier, it will produce a "realistic" sound field, adding a new dimension that could not previously be obtained, in a home listening room.

Since signal processing is all done digitally, there will be no noise or deterioration in the reproduced sound. And, since the RX-1010VTN has a built-in rear-channel amplifier, when a pair of "presence" speakers is added to a normal 2-channel system, the user can easily enjoy the full benefits of the signal processing system. For the user's convenience, seven possible sources — concert hall, etc. — are provided, to produce a "real" sound field without any further adjustment, while adjustable parameters — to compensate for room size, etc. — give the user added flexibility. This system was specifically engineered by JVC for true music lovers, and we're sure it will amaze you. After processing, the resultant sound field is a realistic reproduction of the sound you hear at a live performance.

DOLBY SURROUND

Dolby Surround Sound System with Pro-Logic circuitry*

(RX-1010VTN*, RX-903VBK*, RX-803VBK, RX-703VBK)

This year's top-end receivers feature a Dolby Surround decoder which gives that extra dimension of movie-theater excitement by the perfect reproduction of Dolby Surround-encoded video programs, plus four-channel operation. For the improved clarity of sound effects, we used digital signal processing circuitry in the Dolby Surround decoder circuit.
Furthermore, the RX-1010VTN and RX-903VBK are equipped with "Pro-Logic" circuitry allowing the reproduction of a more realistic sound field, closer to ever the sound in a movie theater.

By simply adding a pair of surround-channel speakers, you can enjoy the real excitement of a three-dimensional Dolby Surround sound field that you previous experience only in a movie theater, in the warmth of your home listening room. Since all decoding processing is performed digitally, no signal deterioration or distortion will be added; only the sound effects intended by the movie's producer will be heard from the rear/surround speakers.

Refer to "Technical Notes" page 13 for more information.

Built-in four-channel amplifier

(RX-1010VTN, RX-903VBK, RX-803VBK, RX-703VBK)

Our top receivers come with a built-in four-channel power amplifier. With this, as well as using the amplifier for normal 2-channel stereo reproduction with dynamic power, you can also use it for four-channel "surround sound" reproduction with a room-filling power output.

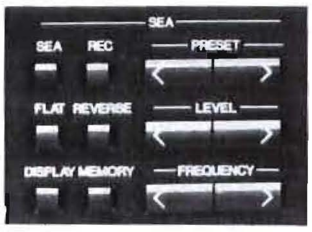
Since receivers with a four-channel capability also incorporate Digital Acoustics and Dolby Surround processors, the user can enjoy surround sound and digitally processed sound fields by simply adding a pair of rear/surround speakers; this is especially effective for recordings of live performances, movie videos, etc.

CCS (COMPU LINK Communication System) interactive

(All models except RX-302BK and RX-201BK)

This CCS system gives further interactive information to the JVC-exclusive COMPU LINK system, by indicating the current source or operation mode you've commanded, or volume up/down, even the station name and SEA pattern recalled.

With CCS system, when you remotely operate the component connected to the receiver, such as CD player or cassette deck, the specified operation in response to the command you've given will be indicated on the FL panel, with a graphic symbol. You can confirm whether you've given a correct command or not at a glance, while the large-sized easy-to-comprehend amber-colored FL display improves the visibility so can be confirmed from your listening position.



Five programmed and five user preset equalization patterns (RX-1010VTN, RX-903VBK, RX-803VBK, RX-703VBK, RX-503BK)

The graphic equalizer not only allows the user to create five custom equalization patterns and store them in memory for later recall but also lets the user call up one of five pre-programmed equalization patterns (HEAVY, CLEAR, SOFT, MOVIE, and VOCAL).

Ten preset equalization patterns can be recalled at the touch of a button; they can then be modified to create any required response.

SEA equalization pattern name memory/indication (RX-1010VTN, RX-903VBK, RX-803VBK, RX-703VBK, RX-503BK)

Together with the five user-programmable equalizer patterns, top-end receivers have a pattern name input and display capability. In the same way as that for station names, up to five or four (RX-503BK) characters or numerals can be stored in memory.

With the SEA graphic equalizer controls, you can create any desired equalization pattern by pressing the electronically controlled frequency buttons. After this, you can store your own name, according to the pattern you've created. Using names like "JAZZ" or "ROCK", for example, you can input any desired name and store it together with the corresponding pattern. Wherever the user calls up an equalization pattern, he or she can see its name in the display, at a glance.

40 FM/AM random station preset memory (All models)

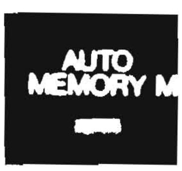
All JVC receivers include a computer-controlled digital synthesizer tuner featuring preset memory of up to 40! broadcast stations for FM and AM randomly. This feature lets the user to preset almost any FM or AM station to be listened to, for immediate pushbutton recall.

Since the tuning condition is controlled by the built-in microcomputer and stored in memory, reception will always be in the best possible condition. Therefore, whenever the required preset channel is recalled, the corresponding station will be recalled with optimum reception. Since up to 40 preset channels are available, you can store almost all stations in your area, now or in future. The 40-channel preset memory will accept any desired combination of FM and AM stations for added convenience.

Station name memory with alphanumeric display (RX-1010VTN, RX-903VBK, RX-803VBK, RX-703VBK, RX-503BK)

By incorporating a large-capacity memory device, up to five or four (RX-503BK) alphanumeric characters can be stored in memory to be used for "station names", etc. The "station name" from storage will be displayed together with the broadcast frequency of the recalled preset channel.

With this convenient feature, you don't have to remember the frequencies of broadcasts, or their preset channel numbers. When you recall a channel, you can check its station name from the character display. You can give preset stations any names you want — "JAZZ" or "ROCK", etc. This gives extra convenience for tuner reception, as well as adding a new dimension to audio enjoyment.



Auto memory functions (All models)

Auto memory operates by using the computer to automatically preset stations one after another, with the user selecting the starting frequency, even in the middle of the tuning band.

This function is extremely convenient, presetting stations in optimum condition wherever you live and eliminating the need to know specific FM or AM frequencies.

Preset scan function (All models)

Preset scan function lets the user automatically tune to each of the preset FM (or AM) stations in memory one by one and hear each for five seconds.

This is a very convenient way to search for a station you want to hear or to sample all the programming offered by your preset stations at any moment.

Sound selector (RX-1010VTN, RX-903VBK)

The Sound Selector lets you combine a picture from any video source with sound from the source you want to hear.

This integrates audio and video the way the user wants it. For example, you can watch a video signal from a VCR and listen to a compact disc, or you can record your favorite sound on a video tape you have recorded using your video camera.

CSRP (COMPU LINK Source Related Preset) System

Sophistication in receiver design

Conventionally, when a user sets a receiver's controls — volume, balance, tone controls, etc. — he or she would use the same settings when playing any program source. But these sources really require different settings, because of differences in level between FM and AM broadcasts, compact discs and tapes, and the different degrees of compensation required with each source.

With the CSRP function of our top-end receivers, when you switch from one source to another, the optimum settings for the new source are recalled from memory. In this way, optimum sound is assured whichever source you are listening to, even when you switch from one source to another.

Basically, everything that is set using the front panel controls is stored in memory for each source independently; with these receivers, this means the volume, balance, the settings of the S.E.A. graphic equalizer, graphic equalization on/off, loudness on/off and surround sound on/off, etc.

The biggest advantage of the CSRP system is that the optimum values for each of the parameters are set instantly the source or preset station is selected. And after they have been set, they are displayed in sequence in the large FL display. Also, when the CSRP DISPLAY button provided on the remote control unit is pressed, the preset values are displayed in sequence for checking from a distance. If necessary, you can change the values using the remote control.

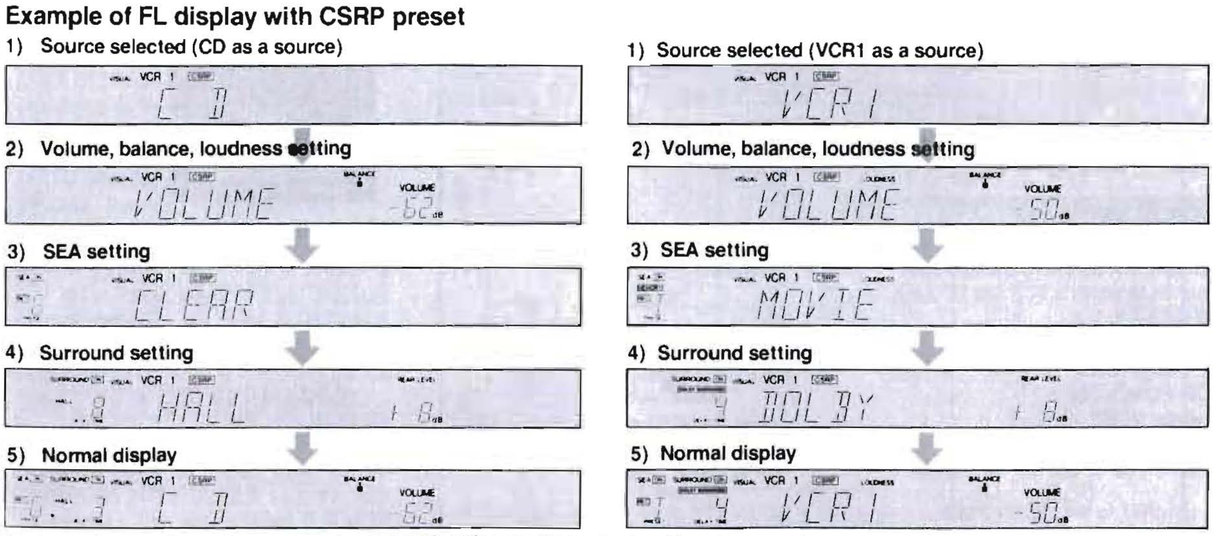
In addition, as the factory-preset values are stored in memory in these receivers, these can be used as they are without any parameter settings when the receivers are switched on for the first time and then modified as required. Also, the CSRP test (demonstration) function is provided with the RX-1010VTN and RX-903VBK so that the settings including the factory-preset values can be displayed in sequence in the FL display until this function is released. This function is effective for an in-store display while users will find it useful for checking their settings for each source in sequence.

The drawbacks of conventional systems

The fact is that each source requires different settings of a variety of parameters. One example is the different levels at which phonograph records and compact discs are recorded. Another problem with these sources is that, while compact discs have a relatively flat characteristic and require less compensation by S.E.A., phonograph records usually require equalization because of the low-frequency resonance point of the cartridge. And different preset stations, even in the same frequency band, could require different settings, because of differences in the signals received from them and even the equipment they use in their studios.

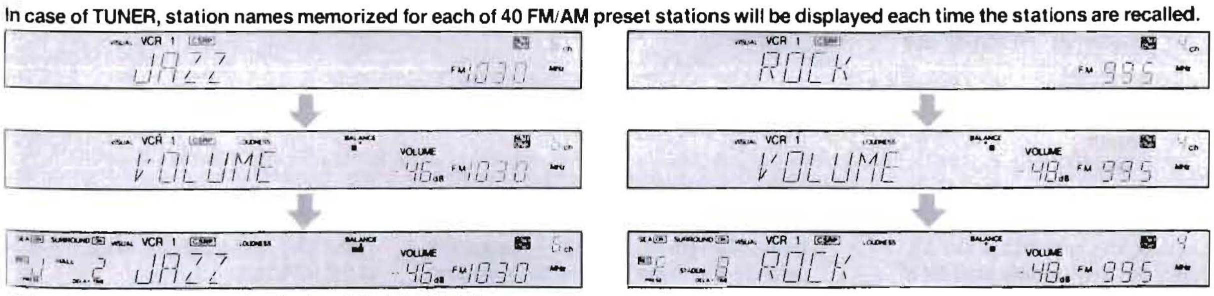
Therefore, to optimize the sound when you change sources, many settings have to be adjusted, not just the volume, but the S.E.A. settings and the values of the various parameters used by the surround sound circuit.

Example of FL display with CSRP preset



As shown above, each time the source is changed, each setting condition is displayed for a few seconds in sequence.

In case of TUNER, station names memorized for each of 40 FM/AM preset stations will be displayed each time the stations are recalled.



* The above example shows the display of the RX-803VBK.

Versatile AV Remote Control, Ideal for Use in the Ultimate AV System

I. AV Remote Control Unit

Unified AV control can be achieved with the RM-SR703U provided with the RX-703BK, the RX-503BK's RM-SR503U and the RX-403BK's RM-SR403U. With these remote controls, you to control not only the receiver itself, but also COMPU LINK Components such as a CD player, cassette deck, turntable, etc. Certain JVC VCRs and TVs can also be remote controlled, from your listening/viewing position. In this way, the remote controls make these receivers ideal for use at the center of a sophisticated AV system.

II. Programmable AV Remote Control Unit

The RM-SR1010U is an advanced LCD programmable AV remote control provided with our top-end RX-1010VTN. In addition to its basic AV remote control functions, a "learning" function is added; this can be used to teach it the functions of other JVC units as well as components from other manufacturers. These learned functions can be recalled at any time, allowing an entire AV system to be controlled from a single remote control unit.

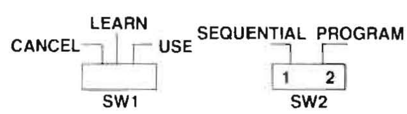
As its comprehensive LCD panel display is switched each time the source is changed and clearly shows each function of the source selected, you can access each function in easy stages without making any mistakes.

III. New Programmable Remote Control Unit

In addition to the functions of an ordinary programmable AV remote control unit, our new remote control unit has another advantage — its sequential programming facility. With this function, you can store up to 6 sequences of operations in memory, with each sequence consisting of up to 16 key operations. After it's been programmed, a complete sequence of operations can be started by touching of single button. This function is provided with the RX-903VBK's RM-SR903U and the RX-803VBK's RM-SR803U. Instead using an LCD display, the keys are arranged on panels with different colors according to their functions and the keys on the CONTROL PANEL SECTION can be used for each source selected in the same way as using the LCD panel. This makes it easier to assign the required functions with the remote control, for enhanced operability.

Sequential Memory Function

On the upper section of these remote control units, there are a switch (SW1) with CANCEL, LEARN and USE positions, a SEQUENTIAL PROGRAM switch (SW2) with positions 1 and 2 and SEQUENTIAL PROGRAM KEYs A, B and C.



How to learn

When "learning" procedures, for example, storing the following steps required to watch a VCR into SEQUENTIAL PROGRAM KEY A.

Audio Power ON
TV Power ON
VCR Power ON
Source VCR1
VCR PLAY
(when a JVC TV and VCR are used)

1) Set SW1 to the LEARN mode.
2) Set SW2 to 1.
3) Keep the SEQUENTIAL PROGRAM KEY A pressed.
   SEQUENTIAL PROGRAM KEY
   [A]
4) Keep the POWER button of the AUDIO pressed.
   POWER
   [AUDIO]
5) Keep the the POWER button of TV pressed.
   [TV]
6) Keep the POWER button of VCR pressed.
   [VCR]
7) Keep the VCR1 of the SOURCE SELECT pressed.
   SOURCE SELECT
   VCR1
   [ ]
8) Keep the Play button of the PROGRAMMABLE CONTROL SECTION pressed.
   PLAY
   [ ]
9) Set SW1 to the USE mode.

Now, the procedures set in steps 4 to 8 are stored in the memory of key A with the SEQUENTIAL PROGRAM switch set to 1. When SW1 is set to the USE mode and the SEQUENTIAL PROGRAM switch is set to 1, just press key A to perform procedures 4 to 8.

Keys B and C can be programmed in the same way. And after setting the SEQUENTIAL PROGRAM switch to 2, another three sequences can be stored in the memories of SEQUENTIAL PROGRAM KEYs A, B and C. In this way, a total of 6 sequences can be stored in memory.

How to cancel

In the above sequence, if key A's memory has been already used for a sequence of operations, when you reach step 3, the LED ERROR on the remote control will light to warn you. To cancel the data key A's memory, follow the procedure below.

1) Set SW1 to the CANCEL mode.
2) Set SW2 to 1.
3) Press SEQUENTIAL PROGRAM KEY A for approx. 2 seconds.

CD PLAYERS

Feature Highlights of '90 CD Players

1

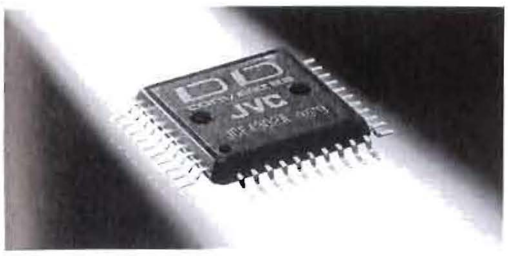
PEM DD** (Pulse Edge Modulation Differential-Linearity Errorless D/A)* ***Converter; no zero-crossing distortion and improved linearity at low levels

2

JVC-original "K2 Interface" coding transmission circuit completely eliminates non-coded components in D/A conversion

3

New DDRP** (Dynamics Detection Recording Processor);* ***synchronized operation of CD players and cassette decks at the optimum recording level, resulting in one-touch recording with a wider dynamic range



PEM DD converter consisting of a fourth-order noise shaper and high-resolution PEM D/A converters

As it uses 1-bit D A conversion, the drawbacks that could not be solved with conventional D/A converters have been overcome. First, zero-crossing distortion and glitches do not occur due to the circuit's principles of operation. Second, as the timing is controlled by a quartz crystal oscillator, operation is extremely accurate. Third, it is not affected by changes in temperature and aging. A further advantage is that, as data values are extracted using the positions of the edges of pulses, the PEM D/A converter has a resolution more than twice that of a conventional 1-bit D A converter. With this PEM D/A converter and the fourth-order noise shaper, requantization noise is reduced to a completely negligible level. As a result, the "presence" of sound field and musical nuances can be reproduced with extreme fidelity.

K2 INTERFACE

New "code transmission system" eliminates jitter and ripple contained in digital signals

The "K2 Interface" technology is an entirely new method to transmit the digital data so that only the "encoded" data is supplied to the subsequent D A converter, etc. With this system, ripple and jitter which cannot be eliminated after D A conversion and which could affect the resultant analog signal are prevented from reaching the analog signal processing section. With the K2 Interface, the music signals are heard without any effects introduced by the equipment or cables used.

The convenient DDRP recording system sets the optimum recording level automatically

The DDRP function makes use of the COMPU LINK Control System to permit synchronized operation of a CD player and cassette deck, automatically optimizing the recording level, for the widest possible dynamic range. The DDRP function lets a CD player (XL-Z431BK) perform "peak search" to determine the optimum recording level. After the "peak search" has been completed, the cassette deck automatically enters the recording standby mode with the input level set to the optimum level for the type of tape used. On a cue from the CD player, the cassette deck switches to the record mode. In this way, high-quality recording can be performed automatically, more easily than with manual level adjustment.

Feature Comparison Chart

CD Players
(Single-Disc Models)

Feature		XL-Z1010TN	XL-Z611BK	XL-Z431BK	XL-Z331BK	XL-V231BK	XL-V131BK	XL-G512NBK (CD+G player)
Digital								
K2 Interface		✓						
Digital Filter	**8fs**	✓		✓	✓	✓		
	4fs		✓				✓	✓
D/A Converter	**PEM DAC**			✓	✓	✓		
	18-bit Combination 4 DAC	✓						
	Dual 18-bit N.S DAC		✓					✓
	Dual 16-bit DAC						✓	
Digital Out	Optical	✓	✓					
	Coaxial	✓	✓	✓	✓			
Mechanism/Servo								
Large Stabilized Disc Clamper		✓	✓	✓	✓	✓	✓	✓
8-cm (3-inch) Disc Compatible Tray		✓	✓	✓	✓	✓	✓	✓
New ISS		✓	✓	✓	✓	✓	✓	✓
3-Beam Laser Pickup		✓	✓	✓	✓	✓	✓	✓
New Y-Servo		✓	✓	✓	✓	✓	✓	✓
Remote Control								
Included		✓	✓	✓	✓			✓
	Volume	✓	✓	✓				
	Display ON/OFF	✓	✓	✓				
	Numeric Keys	✓	✓	✓	✓			✓
Display								
Multi-Function FL		✓	✓	✓	✓	✓	✓	✓
Title Display	Disc		✓					
	Tunes		✓					
No. of Program Chart Tracks		20	20	20	20	20	15	20
Function								
DDRP				✓				
Editing	**3-Way**	✓	✓	✓	✓	✓	✓ (2-Way)	✓
	Side A/B	✓	✓	✓	✓	✓	✓	✓
Memory	Disc/Tune Title Memory		✓					
	Disc Program Memory		✓					
No. of Programs		32	32	32	32	32	32	32
Numeric Keypad (1-10, +10)		✓	✓	✓				✓
Intro Scan (Remote)		✓		✓				✓
Index Search (Remote)		✓	✓	✓	✓			
Random Play			✓					
Repeat	All/Single Track	✓	✓	✓	✓	✓	✓	✓
	A-B (Block)	✓	✓	✓				
	Program Repeat	✓	✓	✓	✓	✓	✓	✓
Search (Auto/Manual)		✓	✓	✓	✓	✓	✓	✓
Output								
Headphone Output	with Volume	✓	✓	✓				✓
	Fixed				✓	✓	✓	
Analog Output	Fixed & Variable	✓	✓	✓				
	Fixed				✓	✓	✓	✓
S-Video Output								✓
MIDI Output								✓
Flip-down Door		✓	✓					
COMPU LINK Component		✓	✓	✓	✓	✓	✓	✓

Feature Comparison Chart

CD Players
(Auto-Changer Models)

		XL-M701BK	XL-M403BK	XL-M303BK	XL-R202BK
Mechanism/Servo					
Type		6-Disc Magazine Plus-One Tray	6-Disc Magazine Plus One Tray	6-Disc Magazine	5-Disc Carousel
3-Beam Laser Pickup		✓	✓	✓	✓
New ISS		✓	✓	✓	✓
New Y-Servo		✓	✓	✓	✓
Digital					
Digital Filter	**8fs**		✓		
	4fs	✓		✓	✓
DA Converter	**PEM DAC**		✓		
	Dual 18-bit N.S. DAC	✓		✓	
	Dual 16-bit DAC				✓
Remote Control					
Included		✓	✓		
	Disc Keys	7	7		
	Numeric Keys	✓	✓		
Display					
Multi-Function FL		✓	✓	✓	✓
Title Display	Disc	✓			
	Magazine	✓			
Function					
Play mode	Continue	✓	✓	✓	✓
	Program	✓	✓	✓	✓
	Random	✓	✓	✓	✓
Editing	**2-Way**		✓	✓	
	Side A/B		✓	✓	
Memory	Magazine/Disc Title Memory	✓			
	Magazine Program Memory	✓			
No. of Programs		32	32	32	32
Disc Keys		7	7	1	5
Numeric Keypad (1-10, +10)		✓			
Repeat	All/Single Track	✓	✓	✓	✓
	Program Repeat	✓	✓	✓	✓
Search (Auto/Manual)		✓	✓	✓	✓
Timer Play		✓			
Output					
Headphone Output	with Volume	✓			
	Fixed		✓	✓	✓
COMPU LINK Component		✓	✓	✓	✓

"K2 Interface" CD player with quadruple full-time 18-bit combination D/A converters

Titanium-finished

XL-Z1010TN SUPER DIGIFINE COMPU LINK Component

Compact Disc Player

- K2 Interface
- 8-times oversampling digital filter
- Quadruple full-time linear 18-bit combination D/A converters
- Optical and coaxial digital outputs
- 3-way editing function

CD player with optical digital output and disc memory

XL-Z611BK DIGIFINE COMPU LINK Component

Compact Disc Player

- Dual 18-bit noise-shaping D/A converter
- 4-times oversampling digital filter
- Disc/track title memory and disc program memory
- Optical and coaxial digital outputs
- 3-way editing function

NEW *CD player with newly-developed PEM DD converter and DDRP function*

XL-Z431BK DIGIFINE COMPU LINK Component

Compact Disc Player

- PEM DD converter
- 8-times oversampling digital filter
- New DDRP
- Coaxial digital output
- 3-way editing function

NEW CD player with newly-developed PEM DD converter and digital output

XL-Z331BK DIGIFINE COMPU LINK Component

Compact Disc Player

- PEM DD converter
- 8-times oversampling digital filter
- Coaxial digital output
- 3-way editing function
- Multi-function display with 20-track program chart
- Remote control

NEW CD player with newly-developed PEM DD converter and 3-way edit function

XL-V231BK DIGIFINE COMPU LINK Component

Compact Disc Player

- PEM DD converter
- 8-times oversampling digital filter
- 3-way editing function
- Multi-function display with 20-track program chart
- Headphone output

NEW Basic CD player with 15-track program chart display

XL-V131BK DIGIFINE COMPU LINK Component

Compact Disc Player

- 4-times oversampling digital filter
- Dual 16-bit D/A converter
- 2-way editing function
- Multi-function display with 15-track program chart
- Random access programming of up to 32 tracks

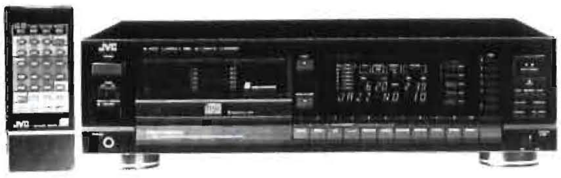
"CD+G" player with 3-way editing function

XL-G512NBK DIGIFINE COMPU LINK Component

Compact Disc + Graphics Player

- Dual 18-bit noise-shaping D/A converter
- 4-times oversampling digital filter
- Built-in graphics decoder
- 3-way editing function
- Multi-function display with 20-track program chart
- Remote control with numeric keypad
- S-video terminal

"6+1" CD auto-changer with magazine/disc title memory and magazine program memo



XL-M701BK DIGIFINE COMPU LINK Component

"6+1" Compact Disc Auto Changer

- Dual 18-bit noise-shaping D/A converter
- 4-times oversampling digital filter
- Remote control with 6 disc keys and numeric keypad
- Coaxial digital output
- Magazine/disc title memory for 11 magazines

NEW *"6+1" CD auto-changer with newly-developed PEM DD converter and remote control*

XL-M403BK DIGIFINE COMPU LINK Component

"6+1" Compact Disc Auto Changer

- PEM DD converter
- 8-times oversampling digital filter
- Remote control with 7 disc keys
- 2-way magazine editing function
- Random access programming of up to 32 steps

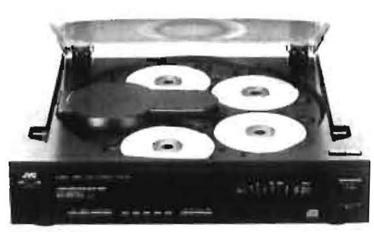
NEW *CD auto-changer with 6-disc magazine and 2-way magazine editing*

XL-M303BK DIGIFINE COMPU LINK Component

Compact Disc Auto Changer

- 4-times oversampling digital filter
- Dual 18-bit noise-shaping D/A converter
- 2-way magazine editing function
- Random access programming of up to 32 steps
- 4-way repeat

NEW *New carousel type CD auto-changer with multi-function display*

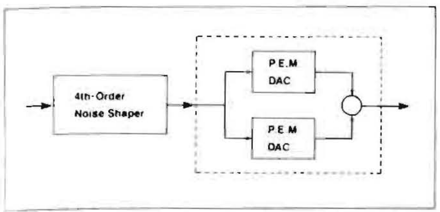


XL-R202BK DIGIFINE COMPU LINK Component

Carousel 5-Disc Auto Changer

- 4-times oversampling digital filter
- Dual 16-bit D/A converters
- Continuous play, random play and program play
- Multi-function display
- Random access programming of up to 32 steps

Feature reference / Technology/Function — Benefits

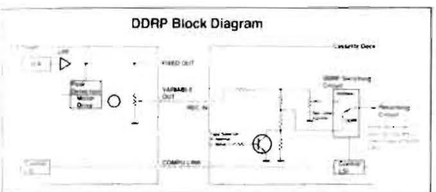


PEM DD converter (XL-Z431BK, XL-Z331BK, XL-V231BK, XL-M403BK)

The PEM DD (Pulse Edge Modulation Differential Linearity-Errorless D/A) converter which uses a fourth-order noise shaper and PEM DACs is a JVC original. The fourth-order noise shaper reduces requantization noise to a negligible level while two PEM DACs increase the resolution to more than twice that of a conventional 1-bit DA converter because data is extracted using the edges of pulses, to output two streams of data.

Refer to "New Hi-Fi Technology 1990" at the end of this book.

As this performs 1-bit operation with a single amplitude and it does not rely on the accurate alignment of the weighted values of a ladder of resistance elements, it is theoretically free from zero-crossing distortion as well as non-linearity at low levels. Also, for accuracy, the switching timing is controlled by the high-precision clock generated by a quartz crystal oscillator; distortion in the amplitude direction cannot occur and it is not affected by changes in temperature and aging. As a fourth-order noise shaper and high-resolution PEM DACs are used, very low signals can be reproduced very accurately so as to reproduce the "presence" of a sound field and musical nuances with extreme fidelity.



DDRP (Dynamics Detection Recording Processor) function (XL-Z431BK)

This function makes use of the COMPU LINK Control System to permit synchronized operation of a CD player and cassette deck, automatically optimizing recording. By simply pressing a button, the CD player performs "peak search" by scanning the compact disc at high speed to determine the peak signal level, so the CD can be recorded without distortion. Then, if the cassette deck to which it is connected is provided with the DDRP function, the deck enters the recording standby mode, judges the optimum recording level for the type of tape used, and the cassette deck starts recording.

Normally, when recording a compact disc on cassette tape, the adjustment of the recording level is complicated and can take a long time. However, with our new system, level adjustment is performed by simply pressing a button on the CD player, after which recording is performed automatically, at the optimum recording level. Therefore, high-quality recording is performed automatically and easily, with the widest possible dynamic range.

Refer to "Technical Notes" page 33 for more information.

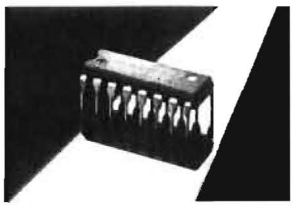
K2 INTERFACE

K2 Interface (XL-Z1010TN)

The entirely original K2 Interface technology was developed in conjunction with software members of the JVC group on the basis of our extensive research and musical-oriented policy. It is not simply a technique to eliminate noise, etc. but to transmit the digital codes which carry the music signal as they are. As its name suggests, it's an "interface" between digital signal processing section and the analog signal processing section, located before the D/A converter. The K2 Interface is an entirely new method to transmit the digital data so that only the "encoded" data is supplied to the subsequent D/A converter, etc. and from there to the analog signal processing circuits.

Unlike conventional circuits which transmit the digital waveforms using an optical isolation technique, or shape the digital waveform after D/A conversion, the K2 Interface incorporated in the XL-Z1010TN, uses an entirely new method to transmit the digital data, called the "code transmission system". In this, only digital "codes" are transmitted, while ripple and jitter contained in the digital signal are ignored. Then, on the other side of the interface, a new set of digital codes are "recreated" before entering the D/A converter. With this system, ripple and jitter which cannot be eliminated after D/A conversion and which could affect resultant analog signal can never be introduced to the analog signal processing section. You can hear "the musical truth" without any effects introduced by the equipment or cables used.

Refer to "Technical Notes" page 27 for more information.



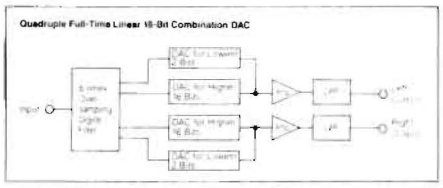
8-times/4-times oversampling digital filter

(8fs: XL-Z1010TN, XL-Z431BK, XL-Z331BK, XL-V231BK, XL-M403BK, 4fs: Other models)

The output from the D/A converter inevitably contains ultrasonic frequencies centered around the sampling frequency (44.1 kHz in CD), spreading from 20 kHz below to 20 kHz above this frequency. Although they are inaudible, the effect they create is heard as noise or distortion of the reproduced sound. To reduce this, a high-order analog filter is used prior to the D/A converter in conventional players. With the analog filter, however, it is difficult to eliminate the above noise components without degrading the sound; it is more effective to use a digital filter to reduce noise while it is in digital form. For further improvement, oversampling is employed in the digital filter, at a multiple of the sampling frequency.

Since the sampling frequency is multiplied by four or eight in models equipped with a 4fs (176.4 kHz) or 8fs (352.8 kHz) oversampling digital filter, the range of the spurious (or aliasing) frequencies before digital filtering is raised far above the range of audible frequencies, for even greater resolution. Therefore, any undesirable signals are completely eliminated in the digital stage, and remaining spurious signals can be removed by an analog filter with more moderate characteristics which does not affect audible frequencies, so that extremely pure audio signals are reproduced.

Quadruple Full-Time Linear 18-Bit Combination DAC

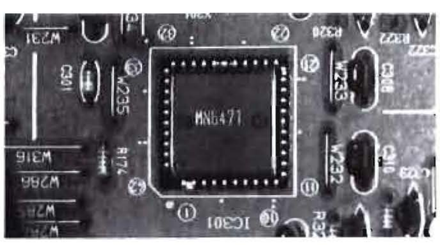


"Full-time" 18-bit combination 4 D/A converters (XL-Z1010TN)

While the sampling rate (44.1 kHz) is multiplied by eight by the digital filter, the number of bits is also raised from the original 16 to 18, and the upper 16 bits and lower 2 bits are processed separately by four "combination" D/A converters, which are always in 18-bit operation, for both the L and R channels independently. This circuit consists of a D/A converter for the upper 16 bits and another for the lower 2 bits, in each channel; these four D/A converters use a combination of LSIs and carefully selected discrete components for the highest possible accuracy.

Unlike conventional "bit-shift" or "floating" system, in which 18-bit D/A conversion is performed only when low level signals are input, our "full- time" system always functions in the 18-bit mode, for greatly improved linearity. And since the lower 2 bits among the 18 data bits from the 18-bit digital filter are processed separately from the upper 16 bits for each channel, the resolution of these two bits which correspond to very low-level signals is greatly improved. Thus the resulting sound can be reproduced with excellent linearity, especially when very low-level passages are played back.

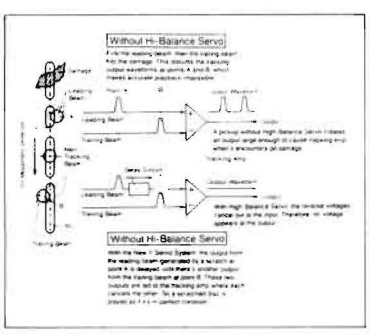
Refer to "Technical Notes" page 32 for more information.



Dual 18-bit noise-shaping D/A converter (XL-Z611BK, XL-G512NBK, XL-M303BK)

This D/A conversion system is realized by the combination of a noise shaper and a PWM D/A converter. After 4-times oversampling by the digital filter, the signal is input to the noise shaper where the signal is further oversampled. This digital data is input to two PWM D/A converters, independently for the L and R channels. These two PWM D/A converters change the data stream into pulses with different pulse widths, which can easily be demodulated into analog signals by the low-pass filter.

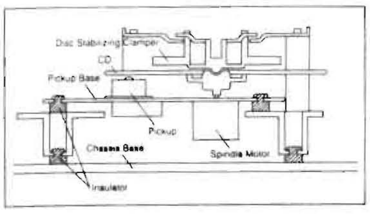
In this system, there's no error on the amplitude axis of the digital waveform while the accuracy of the time axis is maintained by a crystal oscillator. Since D/A conversion is performed by 1-bit operation, there are no differential linearity errors or zero-crossing distortion. Furthermore, the noise shaper before the D/A converter prevents requantization noise. With this system, extremely accurate D/A conversion is possible.



High balance servo (new Y servo system) (All models)

The most important servo mechanism in the new Y servo system is the high balance servo which operates to maintain accurate tracking by cancelling the output from scratches and dirt on the disc; these are detected by the leading beam of the laser pickup by adding the complement of the phase output from the following beam.

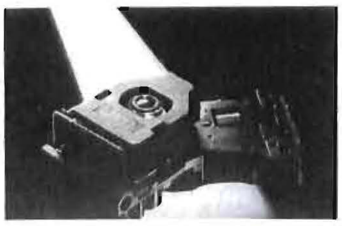
Discs with damaged pits, scratches and dirt, etc. are played back as if they are almost perfect; no mistracking and dropouts occur and digital signals are reproduced with the best characteristics (frequency response, phase response and gain). Therefore, JVC CD players are capable of playing nearly any disc with unerring accuracy and fidelity.



Independent suspension system (All models)

In the independent suspension system, the entire disc drive and laser pickup float "free" from the base to protect them from external influences, such as vibrations, acoustic feedback, etc.

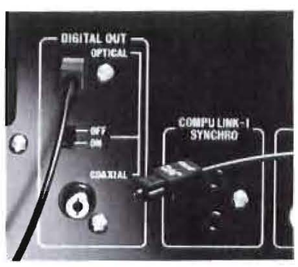
The disc and pickup are protected from vibrations, and the error rate in the reading of digital signals from the discs is reduced. In this way, the overall sound is improved, because less correction of the digital signals is required.



Newly-developed high-precision 3-beam laser pickup (All models)

All JVC CD players are equipped with this new optical pickup, which has been designed to be light and compact while its sensitivity is much greater than that of conventional pickups. The pickup actuator employs a newly developed suspension system, in place of the conventional bearing system. At the same time, the optical path length of the new pickup is almost halved when compared to conventional pickups.

With its small size and lightweight design, the driving current required by the optical pickup is greatly reduced; thus, the pickup's tracing ability is greatly improved while servo noise is reduced. And through the use of an actuator suspension system instead of a bearing system, high-order resonance is reduced for improved stability, while the pickup movement is made smoother as well as more accurate. Because the optical path length is reduced, the dynamic accuracy of the laser beam is greatly improved. By using our advanced computer assisted design, each component is placed in its optimum position so that the effects of optical interference are minimized. In this way, by re-designing all the precision parts of the laser pickup, the laser pickup of our CD players has greater precision in operation, for improved traceability and sound with greater purity.



Digital outputs with optical fiber connector

(XL-Z1010TN, XL-Z611BK; coaxial only: XL-Z431BK, XL-Z331BK, XL-M701BK)

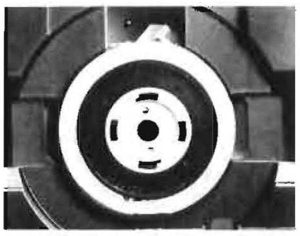
Some CD player models feature a coaxial digital output terminal in addition to conventional analog outputs. This enables digital interfacing with an outboard D/A converter unit or an amplifier with a built-in D/A converter, and thus "pure" signal transmission is made possible.
For an even better performance, the XL-Z1010TN and XL-Z611BK come with an extra "optical" digital out connector which permits digital interfacing using an optical fiber cable for purer signal transmission.

With digital interfacing, since the digital signal picked up from the disc is transmitted as it is directly to the subsequent component without D/A conversion, there is theoretically no deterioration or errors introduced until the signal reaches the subsequent component (amplifier with built-in D/A converter or outboard D/A converter unit). In this way, the extreme purity of the digital signal is maintained.
Optical transmission has another additional benefit; since the signal is transmitted in the form of light via an optical fiber cable, there is no interference from electrical signals (spurious radiation, in technical terms), and no electrical interference between the CD player and the amplifier in which D/A conversion is performed.

8-cm (3-inch) disc compatible disc tray (All single-disc models)

As you know, there's a newcomer in the CD family — the "CD single" with a diameter of 8-cm (3-inch) instead of the 12-cm (5-inch) diameter of standard CDs. Because of their smaller diameters, "CD singles" cannot be loaded directly in some CD players with a sliding disc tray. But all this year's JVC CD players feature a new disc tray which can accommodate 8-cm (3-inch) "CD singles" as well as normal 12-cm (5-inch) discs without a "CD single adapter", etc.

With their smaller 8-cm (3-inch) diameter, "CD singles" provide up to 20 minutes of music compared to the 60 minutes available with regular 12-cm (5-inch) CD discs. Because of their shorter playing time, "CD singles" are most suitable for pops, etc., the type of music that makes the hit parade, while conventional long-playing CDs are used for classical music, etc. All new JVC players can accept these new CD singles without an adapter, as they are.



Disc stabilizing clamper (All single-disc models, XL-R202BK)

This year's CD players employ a large disc stabilizer instead of the conventional disc clamper; this is used to stabilize the moment of inertia of the disc motor, while compensating for the differences in the weight of 12 cm (5-inch) and 8 cm (3-inch) discs.

Since the moment of inertia of 8 cm (3-inch) CD Singles is different from that of 12 cm (5-inch) compact discs, drift of the disc rotation speed differ for the two types of disc, and thus, the servo current required to stabilize the CD Singles will be increased; this may cause slight degradation of the reproduced sound. But by the use of newly developed disc stabilizer, the difference of the moment of inertia between two types of discs is minimized. With this, the servo current required to control the disc rotation is greatly reduced, while the driving torque is therefore increased. As a result, disc rotation is greatly stabilized when either a 8 cm (3-inch) CD Single or 12 cm (5-inch) Compact Disc is loaded.

Disc program memory (XL-Z611BK)

In addition to its disc title and tune title memories, the XL-Z611BK — as a function-oriented model — provides a convenient memory function which stores programmed tracks for each disc, by memorizing the programmed contents together with the TOC (table of contents) data which is then stored in the player's exclusive memory.

Once you program the favorite tunes on each disc, the same programmed contents can be recalled immediately, whenever these discs are loaded. You don't have to re-program every time you load a disc. The player automatically judges the disc by comparing its TOC data (which is encoded at the start of the disc) with the stored data.

Disc/track title memory with alphanumeric display (XL-Z611BK)

With the XL-Z611BK, the user can store the names of discs and tunes in a memory chip installed in the CD player, for all of the discs in your collection. After it's been programmed, whenever a disc is loaded into the player, a 10-digit alphanumeric display will show the title of the disc or tune using the name that you specified, with the memory having space for up to 512 "names".

JVC's exclusive disc memory system lets you store any desired title — the name on the album cover or one of your own — for any CD in your collection. For example, with a compact disc which contains tunes sung by one of your favorite singers, you can give it any "name" that will help you remember what it contains, such as "BEST ALBUM" or "SARAH V #2". Moreover, you can store the titles (or any name you choose) for each tune on the disc in addition to the "disc title". Once this is done, whenever the discs are loaded, the title of the disc or the tune currently being played can be seen immediately without your having to check the credits on the back of the disc case.

Magazine program memory (XL-M701BK)

The XL-M701BK Auto-Changer model permits the user to program up to 32 tracks among the 6 discs in a magazine in any order, for each magazine. And this can be stored in the memory together with the data identifying the magazine.

With this CD changer, once you've programmed your favorite tracks among the six discs in a magazine, this sequence can be held in memory. Therefore, each time that magazine is loaded, it can be played back in the same programmed sequence, automatically every time.

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Magazine/disc title memory with indication (XL-M701BK)

In addition to the convenient "magazine program memory", the XL-M701BK CD Changer model allows the user to program "magazine" titles of up to ten characters for six-disc sets, as well as the "disc" titles of each of the discs contained in a magazine. This convenient function holds the user-specified titles of up to 11 magazines, with six discs in each magazine, in its memory. And these names will be recalled and displayed when the corresponding magazine is loaded.

With this CD Changer model, you get greater flexibility. Because each disc magazine can hold up to six discs, you can "name" a set of six discs "SYMPHONIES" or "LIVE JAZZ", for example, and further, the titles of up to eleven magazines can be stored in memory. Then, whenever you load the disc or magazine, the name you gave it will appear in the 10-digit display. And, while the disc is playing, you can easily check the "title" of the magazine or disc being played back, to see at a glance what's playing.

3-way (2-way) editing function and 2-way magazine editing

(All models except XL-R202BK)

All our CD players feature a versatile 3- or 2-way editing function. "3-way editing" includes auto editing, program editing and multi-disc editing functions while "2-way editing" includes auto editing and program editing. "Auto editing" allows automatic selection of tunes according to the length of the recording you want to make. "Program editing" permits the user to select the tunes to be played in programmed order. And "multi-disc editing" adds further convenience to programmed play, letting the user change discs within the specified program. When recording the program onto tape, the player will tell you to change the disc to record the next programmed tune, whenever necessary. With Auto-Changer models, since "program editing" includes "multi-disc editing", the same editing facility is available even though the feature is called "2-way" editing.

With this highly convenient feature, the user can easily tape his or her favorite tunes from the compact discs in any desired order. With automatic editing, the player automatically determines which tunes should be recorded according to the length of the tape, after you have designated the recording time. And with program editing, you can select the next tune after checking that its playing time does not exceed the time left on the tape, as in programmed playback. Further, with multi-disc editing, the user doesn't have to remember exactly which tune is where or the playing order. Now, "order-made" selections are possible.

Display ON/OFF Switch (XL-Z1010TN, XL-Z611BK, XL-Z431BK)

The XL-Z1010TN, XL-Z611BK and XL-Z431BK are equipped with a "Display ON/OFF" switch which, as you will realize from its name, turns the FL display on and off.

Why would you want to switch off the colorful illuminated display? If you want to listen to extremely low level passages or the "nuances" of music from your discs, any digital noise generated by the FL display (even at very low levels) will be too much to ignore. Since the display can be turned ON/OFF from your listening position using the remote control, you can enjoy music with all its subtlety.

Random access programming (All models)

Random access programming permits programming up to 32 tracks in any order by simply pressing a numeric keypad, for playback in the programmed order.

You can enjoy playback of selections in any desired order by utilizing this quick and easy-to-use programming function.

Random play (XL-Z611BK and all auto-changer models)

This function lets the CD player play tunes selected by the microcomputer in random order.

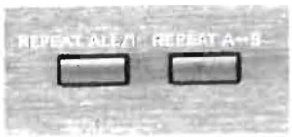
As you never know what tune will be played next, playback of even familiar albums is a new and exciting experience; even when a compact disc is played over and over again, listening is more enjoyable as the order of play is always different. If this function is used with the repeat function, the playing order is changed each time the disc is repeated, to avoid monotony.

Technology/Function | **Benefits**

Intro scan function (XL-Z1010TN, XL-Z431BK)

By simply touching a button, you can listen to the first 15 seconds of every track or all programmed tracks on a disc, one after another.

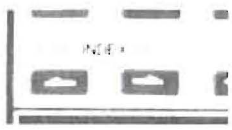
You can easily find the tune you want to hear; when you use this function together with the random play function and repeat function, your listening pleasure will be expanded.



Repeat function (All models)

Five-way repeat is available; all tracks on a disc, a single track, programmed tracks, any segment between any two points and random-play repeat are possible. Different models have different numbers of repeat functions. "4-way" includes all tracks on a disc, a single track, programmed tracks and A-B segment while "3-way" includes all tracks on a disc, a single track and programmed tracks. For more details, refer to "Feature Comparison Chart" on page 16 — 17.

By increasing flexibility of playback, this enhances listening pleasure, letting you enjoy CDs the way you want to; you can also combine this function with the random play and intro scan functions for even greater musical satisfaction.



Index skip (XL-Z1010TN, XL-Z611BK, XL-Z431BK, XL-Z331BK)

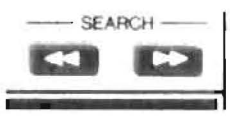
This function is useful to find the desired index point and play from there.

When playing back a long tune with many index points, you can skip to the required index location quickly; this function offers a more precise search capability.

Auto search (All models)

This function moves the pickup to the start of the next, current or previous tune. Every time the button is pressed, the pickup skips forward or backward by one tune.

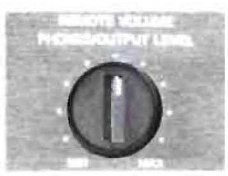
Listening to the same, the next or previous track is easier; this feature lets you search tracks in sequence.



Two-speed manual search (All models)

You can search for any desired section at two speeds while monitoring the speeded-up sound.

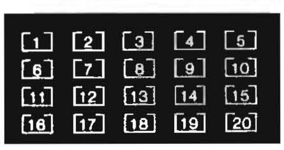
This is a convenient function because you can easily access the approximate position of the selection at high-speed, then exactly the right point with low-speed search.



Motor-driven volume control (XL-Z1010TN, XL-Z611BK, XL-Z431BK)

The volume control is driven by a motor and can be controlled from the remote control unit.

The user can control the volume level from his listening position.



Program chart (All single-disc models)

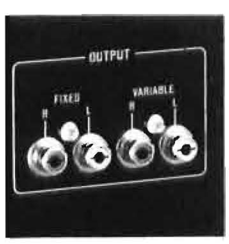
A new style program chart is provided to show at a glance which tracks are being programmed, currently played, or how many tracks there are on a disc.

This helps you to program or select tunes with its easily understandable, logical display.

6 + 1 CD autochanger (XL-M701BK, XL-M403BK)

The JVC autochangers employ an original 6+1 system designed to offer multi-disc CD convenience and long play capability with the ease of loading a single-disc player.

If the lower slot is used, the unit works as a normal CD player and the user can change CDs in the normal way. When the magazine is used with the lower slot, you can enjoy continuous play of up to 7 discs or programmed play of up to 32 selections from a total of 7 discs.



Two analog outputs (XL-Z1010TN, XL-Z611BK, XL-Z431BK)

There are two output modes; fixed and variable.

If the user connects the variable output terminals, the volume can be controlled with the remote control unit within the range of volumes previously set with the amplifier's volume control; the volume (input level) can be set to almost the same level as other sources such as the tuner so that the volume does not change when switching from one source to another. If the fixed output is selected, constant level signals are provided to other components such as a tape deck, etc.

K2 INTERFACE

The K2 Interface used in JVC's top CD player, the XL-Z1010TN, incorporates revolutionary technology to transmit the digital signal by a completely new "code transmission system", instead of the conventional "waveform transmission system".

This newly developed system is not just a technique for eliminating noise like "waveform shaping", but an entirely new method to "recreate" the original digital waveforms, so that components not related to the music signals cannot enter the analog circuitry and therefore cannot introduce distortion. It is totally different from apparently similar systems used by other audio manufacturers; although they claim to achieve results that are similar to those obtained by our "K2 Interface", they are only effective in obtaining clean "voltage waveforms". With the K2 Interface, "current waveforms" are also accurate.

What factors can affect digital sound?

In theory, the quality of digital sound should never change as long as the code information from the source is kept in its original digital form. But there are differences, with different sources, different equipment, even when different cables are used to connect components. Recognizing this, engineers at JVC's R&D Center, cooperating with Victor Musical Industries, the software arm of JVC — to research both the hardware and software aspects of the problem — started to investigate the subject 5 years ago.

We found that "ripple" — waveform distortion — and "jitter" — timing deviation — in the digital signal greatly affected the quality of the sound heard from the speakers. Ripple and jitter are introduced by such factors as fluctuations in the power supply, and stray capacitance and inductance in the circuits. The ripple and jitter are totally unrelated to the music and are therefore "non-code" rather than "code" signal components. Even though they do not change the binary digits in the signal (the 0s and 1s picked up from the compact disc, for example), when they are added to the signal, they produce an effect which can be heard by the listener.

Once these "musically unrelated" components have entered the digital signal, they cannot be removed by an optical transmission system or any other such device. Even worse, if non-code components are transmitted to the analog section, they can reach the power supply line in the form of current and from there can affect the analog signal after D/A conversion, even if it has been "shaped" to obtain a "cleaner" waveform.

Fig. 1 Non-code components

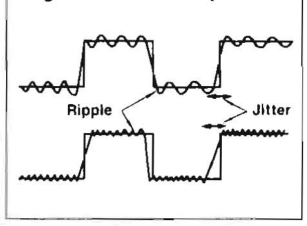


Fig. 2 Concept of conventional digital signal transmission system

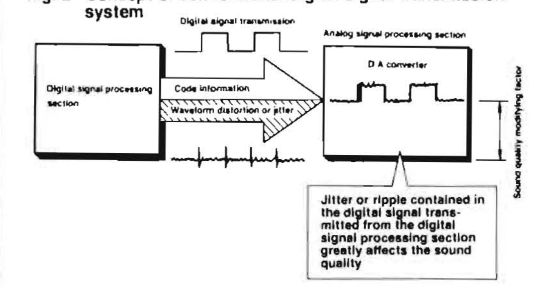
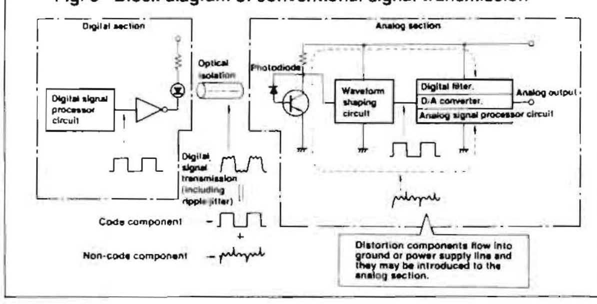


Fig. 3 Block diagram of conventional signal transmission



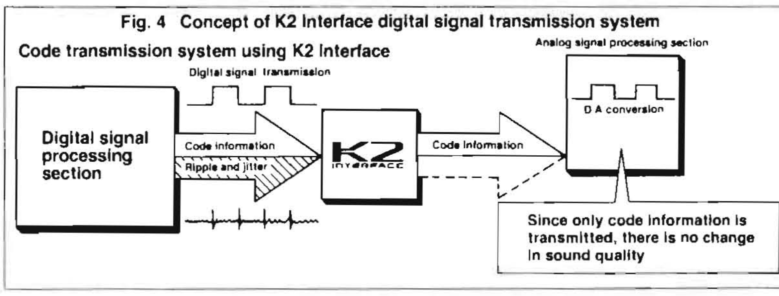
The K2 Interface

To overcome this problem, we developed an entirely new method called a "code transmission system", which replaces the conventional "waveform transmission". Our "K2 Interface" incorporates this revolutionary and totally original new concept.

An interface is the point at which two circuits are connected. In this case, it is where the circuits processing purely digital signals "interface" with the circuits involved in the D/A conversion of this signal and the subsequent analog processing of the music signals. The "K2 Interface" completely eliminates any "non-code" signal components, so that only "code" components, related to music, are D/A converted and amplified.

The K2 Interface consists of two blocks, a Transmitter in the circuit which processes the digital signal, and a Receiver in the "musically related" circuit, immediately before the digital filter and D/A converter.

Fig. 4 Concept of K2 Interface digital signal transmission system



■ How the K2 Interface operates

The signal from the disc after transmission to the processing circuitry, including both code and non-code components, enters the K2 Interface which reads only the code information, consisting of a stream of 0s and 1s. These 0s and 1s are used to recreate an entirely new digital signal on the other side of the K2 Interface, which is used for subsequent processing.

On the digital side, because of the timing signals, the K2 Interface reads only code data. On the analog side, by processing the signal at a very high speed, it is able to discriminate between the 0s and 1s in the code digital data from the disc and the non-code components resulting from ripple and jitter and recreates the stream of digital codes; non-code components are completely eliminated.

So that these non-code components cannot influence the circuitry following the interface, the two stages of the K2 Interface are completely isolated by "photocouplers" in which electrical signals are converted into light, transmitted as pulses of light, then converted back to electrical signals by phototransistors.

There are two photocouplers in the K2 Interface, operating in opposite directions. One is for the transmission of code data signals from the Transmitter to the Receiver, the other is for the transmission of timing signals from the Receiver to the Transmitter.

A crystal oscillator in the Receiver block, on the analog signal processing side of the K2 Interface, is used to generate the sync and timing signals used by the K2 Interface.

The key element of the K2 Interface is the "code detection switch", not included any other "noise eliminating" systems, a switching transistor controlling the operation of the phototransistor on the analog side.

Without this switch, any signals present on the digital side could flow to the analog side. In the K2 Interface, signals can only flow when this code detection switch is closed. This code detection switch is essential for the operation of the K2 Interface and is the key component of the interface.

Fig. 5 K2 Interface Block Diagram

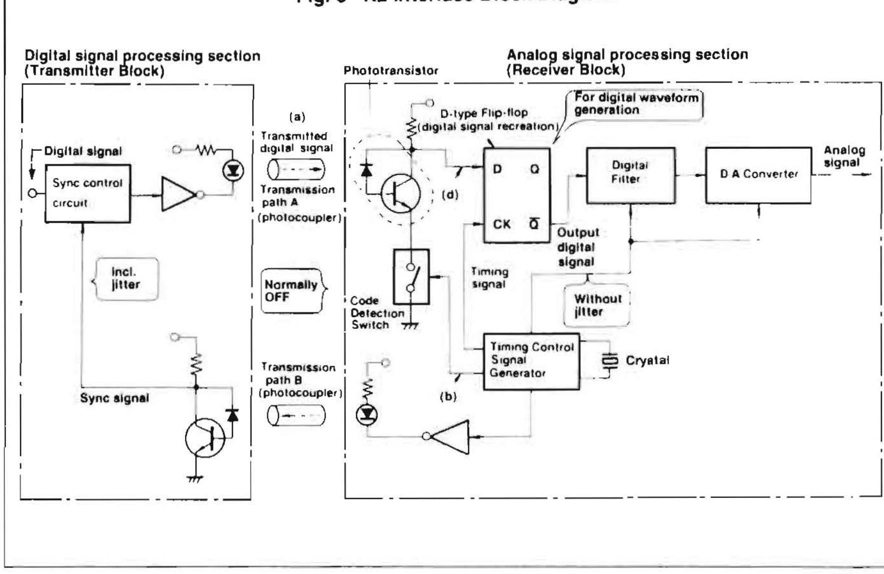
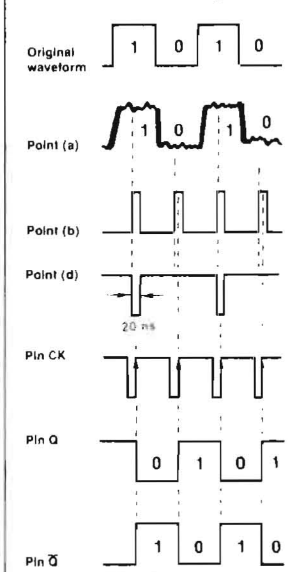


Fig. 6 Operating waveforms at each point



■ Operating concept

The operation of the K2 Interface is controlled by a Timing Control Signal Generator in the Receiver Block. This provides two timing signals, to the code detection switch, and through the Path B photocoupler, to the Transmitter Block.

Following the timing of the signal supplied through the Path B photocoupler, the digital waveform from the disc (in the case of a CD player) is converted into pulses of light in the Transmitter Block, which are supplied to the Receiver Block via the Path A photocoupler.

At the receiver end of the Path A photocoupler, the code detection switch is usually turned OFF; it is turned ON momentarily only when the valid bit could be present, to judge whether the code is a 0 or 1, then it is turned OFF again. Thus, the digital signal from the Transmitter Block flows into Receiver Block only when the code detection switch is turned ON.

An easy to understand analogy is the use of a camera. When photographing a moving object, you use a faster shutter speed. With a shutter speed of 1/1000 or 1/2000 sec., you can get clear and sharply focused images of action.

In a similar way, by reading the digital signal by opening the "gate" for very short periods, only the code information will be transmitted with almost no "non-code" components.

In this way, the signal stream exactly corresponds to the code signals.

After the code detection switch, the signal is applied to a flip-flop in which a new square-wave signal is generated. This newly generated signal will be exactly the same as that picked up from the source (disc); from the flip-flop it is applied to the digital filter, digital-to-analog converter and subsequent audio circuitry.

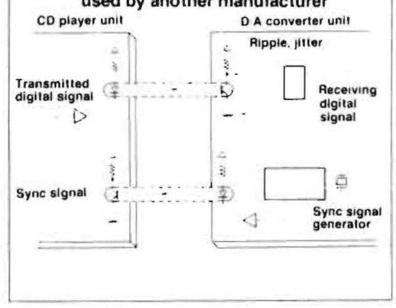
Operations concerning digital audio signals are all done at extremely high speed, with times measured in "nanoseconds" abbreviated as nsec. 1 nsec is one-billionth of a second, 0.000000001 sec.

The time required to read one pit from a compact disc is about 700 nsec. The code detection switch, on the other hand, has a switching time of only 20 nsec. Because its switching time is well within the timing of the digital signal from the disc, it is able to discriminate between code and non-code signal components, thereby eliminating the effects of ripple and jitter.

■ Comparison between K2 Interface and similar circuit from another manufacturer

In the K2 Interface, the transmission block (player and digital signal processing section) and the receiving block (D/A converter and analog signal processing section) are electrically separated, with a timing control circuit (clock generator) located in the receiving block.
Although this circuit layout may look similar to that in a number of CD players from other manufacturers, including the "Twin Link" using photocouplers electrically isolate the two blocks, the key component of the K2 Interface, the code detection switch, is not used by other manufacturers and this make the systems completely different.
In other systems, the digital signals are transmitted as waveforms, and these contain "non-code" components. On the other hand, the K2 Interface transmits only digital data — the stream of 0s and 1s, validated by the timing of operation of the code detection switch — not the digital waveform, and recreates the digital waveform in the receiving block.

Fig. 7 Block diagram of similar circuit used by another manufacturer



■ Another application of the K2 Interface

The K2 Interface, as used in this year's top CD player, the XL-Z1010TN, requires a bi-directional link between the digital section and the analog section which incorporates the clock generator to control the timing with which data is read out from the digital section.

Therefore, this circuit can be used in any digital source component, not only a CD player, so in future it will be used in DAT (Digital Audio Tape) decks and BS (Broadcast Satellite) tuners, etc.

Another slightly different version of the K2 Interface is incorporated in the AX-Z1010TN, the top-end digital reference amplifier which incorporates its own D/A converter, to receive digital signals directly from digital source components.

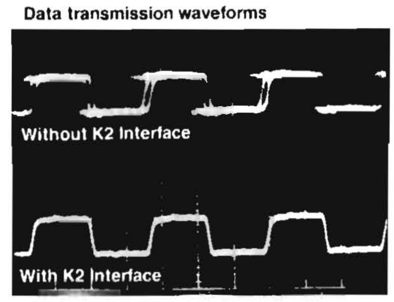
■ The K2 Interface and its advantages

With the K2 Interface in a home hi-fi component system, sound from digital sources has far better resolution, the ambience is greatly enhanced, and the sound field has greater depth and realism than was ever possible before.

Most important of all, with the K2 Interface the sound you hear from a compact disc is exactly what was recorded, reproducing exactly the sound that the musicians, the producer and recording engineer intended. (For use in recording studios, etc., we've already started to develop a "professional version" of K2 Interface.)

There's no magic, and you don't get something that wasn't there originally. Even with the K2 Interface, the signal after D/A conversion is not an improvement on the original digital source signal. Its purpose is to transmit the digital signal from the program source to the analog section without any degradation and without modifying it in any way.

Data transmission waveforms



■ An experiment that proved the effectiveness of the K2 Interface

To prove that our K2 Interface really worked, we performed an experiment, that demonstrated that it really does exclude "non-code" components.

We applied a source generating load variations (distortion resulting from random noise) externally to the digital signal processing section, while a 1 kHz signal was supplied instead of a music signal, using one player equipped with the K2 Interface and one without it, and connected an oscilloscope to the analog outputs, for measurement.

Fig 8 (a) shows the frequency response of the residual noise included in the analog signal. When random noise is applied externally, in the player without the K2 Interface, the noise level is higher and noise has different levels at different frequencies as shown in Fig. 8 (b), which can modify the characteristics of the sound. This shows that distortion applied in the digital stage affects the analog signal, and even worse, the musical contents of the original music signal might be modified depending on how this noise varies.

In the equipment with K2 interface, however, there is no variation or modification of the residual noise, as shown in Fig. 8 (c). That is, the distortion in the digital stage is almost completely eliminated and it never affects the resultant analog signal.

The results of the above experiment show how external noise (distortion) can affect the analog output signal which in turn could affect the purity of the music, and that with the K2 Interface, this external noise in the digital signal is completely excluded, for a resultant analog signal that is exactly the same as the input signal.

(a) (b) (c)

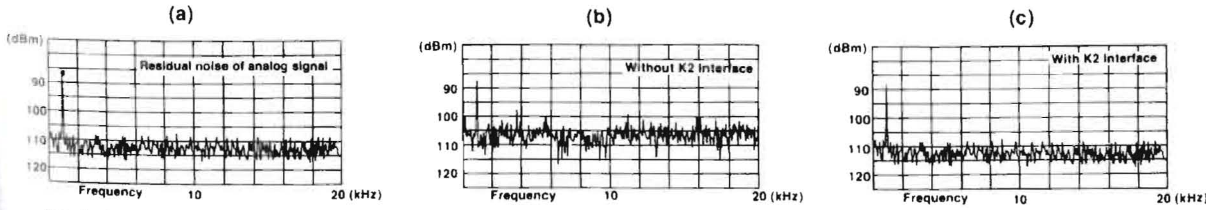


Fig. 8 Frequency spectrum of analog output

JVC Advanced D/A Conversion Technology

Digital sound quality differs greatly depending on the equipment or even the cables used, even when playing the same program, from a compact disc, for example. This is caused by a number of factors including the analog filter after the D/A converter which can affect the resultant analog signal and the leakage of digital signals into the analog section.

But the main circuit in any digital audio component is the D/A converter which converts the digital source signal into analog signals which are then amplified to drive the speakers.

The most effective way to improve the quality of sound obtained from digital sources is to improve the accuracy of D/A conversion, whether it is performed in the digital source component or in an amplifier equipped with its own D/A converter. We have two approaches to improve the accuracy of D/A conversion. One, used in our ladder-type D/A converter, is to improve the resolution of the D/A converter, by sampling at a multiple of the basic sampling rate (oversampling), and to increase the number of bits for more accurate linearity. By using these two techniques, the reproduction of digital source signals can be done more accurately, even with very low level singals. For this purpose, JVC uses a "full-time 18-bit combination quadruple D/A converter".

Another approach is to use a 1-bit D/A converter for the elimination of "zero-crossing distortion" and "non-linear distortion" which degrade the quality of the resultant analog signals. For this purpose, JVC developed a PEM DD (Pulse Edge Modulation Differential Linearity-Errorless D/A) converter.

Before starting to explain the "full-time 18-bit combination quadruple D/A converter" and "PEM DD converter", it would be better to consider the basics of analog-to-digital (A/D) and digital-to-analog (D/A) conversion — the heart of digital audio technology.

■ *How are digital signals changed in D/A conversion?*

Basic A/D conversion

When an analog signal is converted into a digital signal, it is "sampled" at a certain frequency (the sampling frequency) which produces a series of samples with different heights, then these heights are converted into digital values through "quantization". These digital values are binary — a series of 0s and 1s — and it is these that are recorded on the compact disc, etc.

As you can see from Fig. 1, the samples are like steps, the corners of which are outside the waveform of the original analog signal, and these corners result in "quantization" noise.

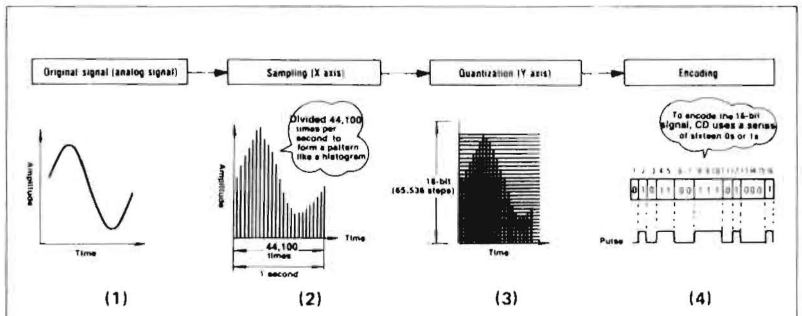


Fig. 1 A/D conversion principle

D/A conversion and low-pass filter

This digital data cannot be heard as it is, and must be re-converted into an analog music signal which can be heard from speakers or headphones. This is done by D/A conversion which reverses the A/D conversion process described above.

When a digital signal is recorded on a compact disc, it's just a series of 0s and 1s. When this data is converted to recreate an analog signal, it produces a stepped waveform as shown in Fig. 2, which, theoretically, is exactly the same as the waveform shown in Fig. 1.

This "stepped" waveform is made up from the smooth music signal (original signal) together with the "corners" related to the sampling frequency, re-quantization noise, corresponding to the quantization noise mentioned earlier.

In the compact disc format, the sampling frequency is 44.1 kHz, that is each second of the music signal (b) is divided into 44,100 slices (waveform (a)). The "corners" which represent quantization noise (which should be eliminated if the original music signal is to be reproduced accurately) are related to timing signal (c) at the sampling frequency.

To reduce quantization noise, the first generation of CD players used an electrical filter called a "low-pass filter" to eliminate unnecessary signals which have frequencies distributed on both sides of the sampling frequency (44.1 kHz, in this case).

As shown in Fig. 3, since the sampling noise is within the range of frequencies from 20 kHz below to 20 kHz above the sampling frequency of 44.1 kHz, a low-pass filter is required to eliminate noise below 24.1 kHz. Because this is close to the upper limit of human hearing, the filter must have steep roll-off characteristics and therefore could easily affect audible frequencies. Early CD players used multi-stage (high-order) analog filters, and this was a major factor in degrading the quality of the resultant music signal in many instances.

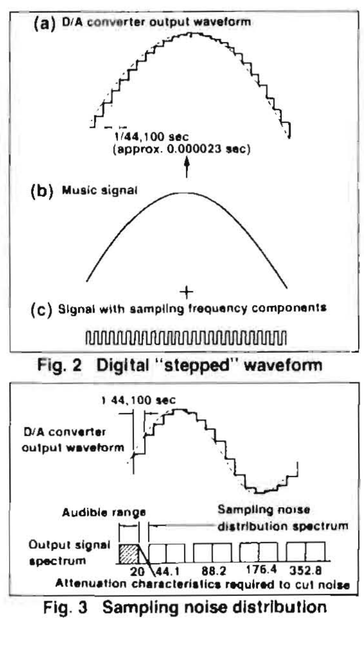


Fig. 2 Digital "stepped" waveform

Fig. 3 Sampling noise distribution

What is "oversampling"?

One technique use to overcome this problem is by "oversampling"; in oversampling, the sampling frequency is raised so that the noise is shifted above the range of audible frequencies, where its influence cannot be heard.

For example, when the sampling frequency is doubled to 88.2 kHz, the lower limit of the sampling noise will be 68.2 kHz, so a low-pass filter with mild roll-off characteristics can be used after the D/A converter.

This posses a new question. How can a digital signal recorded with a sampling frequency of 44.1 kHz be sampled at 88.2 kHz?

Digital signal processing including digital filtering consists of a series of computations, and these arithmetic operations make "oversampling" and other forms of digital processing possible.

Actually, in oversampling, there are blanks between the slices of data sampled at 44.1 kHz, and these blanks are filled in by the digital filter using an arithmetic process called "interpolation".

That is, when the sampling rate is quadrupled to 176.4 kHz, the three slots between the original data slices are filled by interpolation, and when it is multiplied by 8 to 358.8 kHz, seven slots are filled; in both cases the lower limit of sampling noise is raised far above the range of audible frequencies.

Fig. 4 2-times oversampling

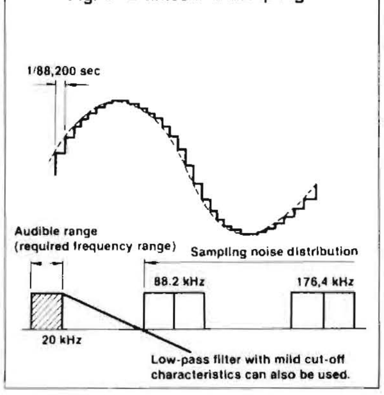


Fig. 5 Oversampling concept

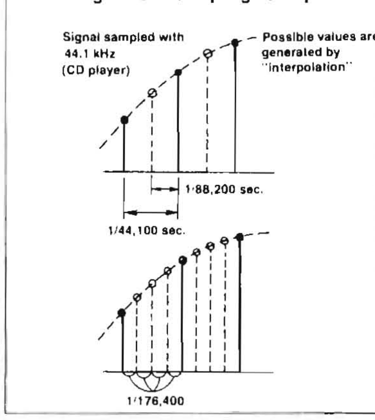
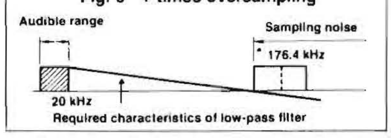


Fig. 6 4-times oversampling



What is a higher bit rate?

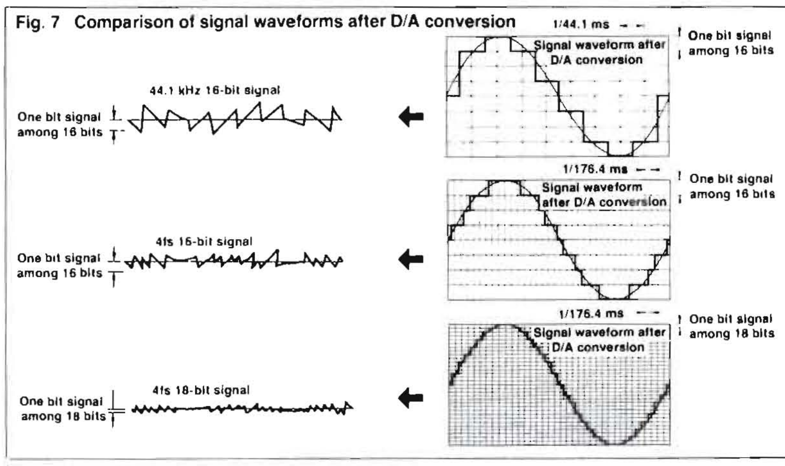
As described above, "oversampling" refers to increasing the number of vertical "slices" in a fixed interval; "increasing the bit rate" increases the number of horizontal "slices" of the step-shaped waveform.

After 16-bit D/A conversion, the signal can have any of 16 heights, with one-bit differences So, when the number of bits is increased, the difference in the heights of steps will be smaller, and thus the signal after D/A conversion will be smoother and closer to the original analog signal.

The differences between the original analog waveform and the "step-shaped" digital signal waveform result in "quantization noise" or "quantization distortion".

By raising the sampling rate and increasing the number of bits, as shown in Fig. 7, this quantization noise has been greatly reduced, and thus the waveform after D/A conversion is much closer to the original analog waveform.

Fig. 7 Comparison of signal waveforms after D/A conversion



■ Full-time 18-bit combination 4 D/A converter

Digital signals are intrinsically "angular"; to convert them into smooth, natural analog waveforms, it is necessary to improve the resolution of the digital-to-analog converter. To achieve this, the XL-Z1010TN processes the signal using 18-bit words rather than the 16 bits used when the signal was initially quantized.

The minute variations in signal level which are necessary to reproduce the subtleties of digital sound are all at the limits of the basic waveform; when the waveform is digitized, as they represent small variations in the waveform, they are in the bottom bits of the "words" recorded on the disc. By dividing the words into two parts and by processing these two parts separately, it is possible to reproduce the music with far greater accuracy.

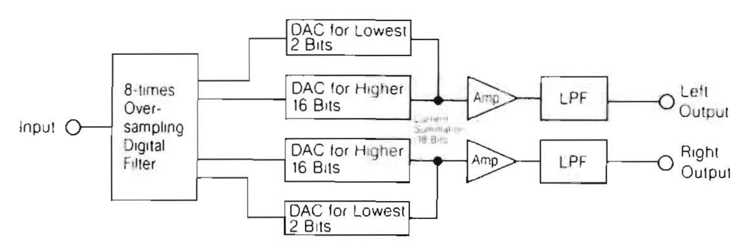
We therefore use two independent D/A converters for each channel, one handling the bottom 2 bits and the other handling the other 16 bits. The electronic components processing the bottom 2 bits which require extreme accuracy are added externally to the existing 16-bit D/A converter (as discrete components, not within a single chip); this greatly improves the resolution of D/A conversion and therefore the fidelity of the sound.

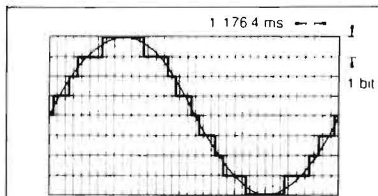
Why extremely low level signals are so important?

In music signal waveforms, small differences are due to variations of pitch and represents the unique "voices" of individual instruments or singers.

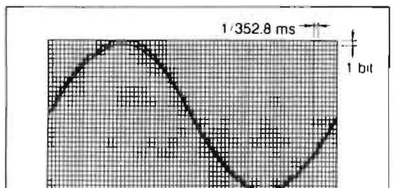
When these waveforms are encoded into digital signals then later decoded to recreate analog waveforms, the minute variations in level may be omitted due to the "stepped" shape of the digital signals. The smaller the width of each "step" and the greater the accuracy with which the height of each step can be judged, the better the quality of the resultant analog music signal. For this, D/A conversion with higher resolution is required.

Quadruple Full-Time Linear 18-Bit Combination DAC





4-times oversampling (16-bit)



8-times oversampling (18-bit)

Some other manufacturers use "bit-shift" or "floating" 18-bit D/A converters, in which a 16-bit D/A converter is used to simulate 18-bit conversion. The 16-bit digital source signal passes through a digital filter, the output of which is shifted to correspond to the signal level; when the level of the music signal is low and there are unused bits at the top of the word, it is shifted up by 2 bits (and the gain reduced by a factor of 4) to improve resolution.

On the other hand, JVC's 18-bit D/A converter has a full 18-bit capacity, and always operates as an 18-bit D/A converter, to obtain greater accuracy in D/A conversion. This is why we call it "full-time" 18-bit operation.

■ New PEM DD converter — another D/A conversion technology from JVC

Even though D A conversion has been greatly improved as mentioned above, there are inevitable problems which are inherent in ladder type D/A converters which use a number of constant current sources, such as zero-crossing distortion and non-linearity distortion at low levels, etc To overcome these problems, JVC's original PEM DD converter was developed using a completely different approach, to improve D/A conversion. As this D/A converter uses 1-bit operation, there is only one signal amplitude and it does not rely on the accurate alignment of the weighted values of a ladder of resistance elements. In addition, as this PEM DD converter consists of a fourth-order noise shaper and two high-resolution PEM DACs, requantization noise is reduced to an insignificant level while resolution is more than twice that of a conventional 1-bit DA converter. For more details, refer to "New Hi-Fi Technology 1990" at the end of this book.

New DDRP (Dynamics Detection Recording Processor) Function

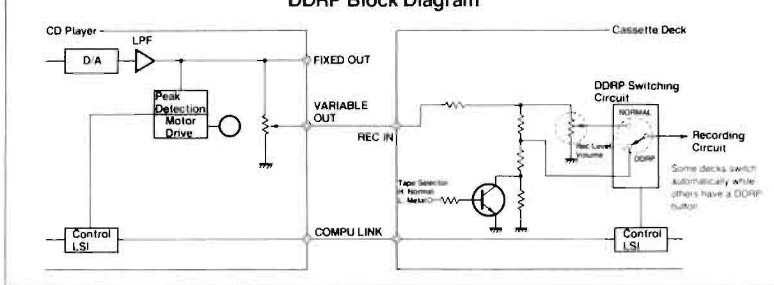
The DDRP function makes use of the COMPU LINK Control System to permit synchronized operation of a CD player and cassette deck, automatically optimizing recording for the widest possible dynamic range. This year, the DDRP function is provided in a new CD player, the XL-Z431BK, and all new cassette decks (except the TD-W103BK).

Operation of the DDRP function

When the DDRP button of the CD player is pressed, DDRP performs "peak search" by scanning the compact disc at high speed, detecting peak music signals to determine the optimum recording level for the particular disc, with the volume control first set to MAX then moved down progressively as higher peaks are detected. The cassette deck provided with the DDRP function is automatically set to the record pause mode and switches the circuit so that signals bypass the input level control circuit for recording with reduced noise while the input level is fixed according to the type of tape (a sensor circuit automatically discriminates between Normal CrO2 and Metal tapes and sets the input level to +3 dB for the former and +6 dB for the latter). Then, on a cue from the CD player, the cassette deck switches to the record mode, and the signal is transferred from the CD player to the cassette deck

Together with the versatile editing system of the CD player, DDRP is the ideal way to make high-quality recordings automatically, more easily than with manual level adjustment.

DDRP Block Diagram

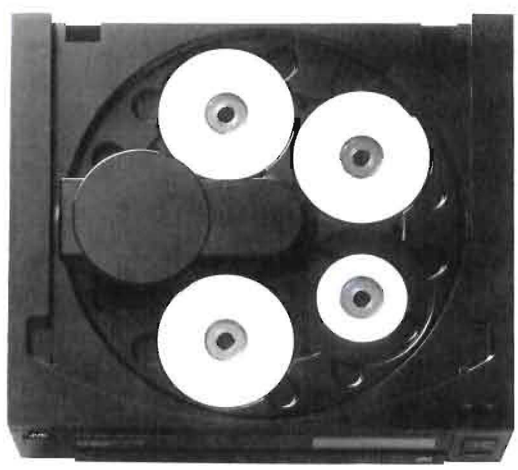


Newly-Introduced Carousel Type CD Auto Changer

This year, for the first time JVC has a carousel type CD auto changer. The XL-R202BK carousel type CD auto changer can hold up to five discs at a time, on a turntable with a transparent plastic cover. In this system, dual 16-bit DACs, a 4-times oversampling digital filter, JVC's high-precision 3-beam laser pickup, a disc stabilizing clamper, JVC's Independent Suspension System and our new Y Servo System are used for quality digital sound and superior tracking ability. This CD auto changer has the following advantages over magazine-type CD auto changers.

1) As all five CDs can be seen through the dust cover, all titles as well as which disc is being played can be checked.
2) Loading discs can be done easily and quickly as they're placed in recesses in the turntable. Even when one CD is being played, the others can be replaced as required.
3) As customers can see a display showing the random play operations — when another CD is selected and moves into position, the circle in the display moves to numbers corresponding to discs that have been loaded in sequence and stops at the number of the next disc to be played, for example — it can be used for an effective in-store demonstration. You can also point out that the turntable rotates in both clockwise and counterclockwise directions, for the fastest access to the required disc.

The major operation features of this CD auto changer are continuous play, random play without repeating the same track, random access programming of up to 32 tracks, 2-way repeat, compatibility with 3-inch CD singles, auto/manual search, etc.



Extremely Flexible CD Editing Facilities

With the wide use of 8-cm (3-inch) CD singles and the increased availability of compact discs, all our CD players (except the XL-R202BK) are provided with a flexible CD editing facility. Most single-tray players have 3-way editing, enabling easy and convenient selection of tunes on compact discs, to be recorded onto cassette tapes. For the same convenience, with our auto-changer models, the user can program any required tracks among the 6 discs contained in each magazine.

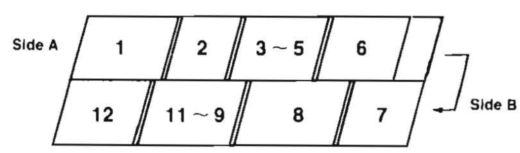
Another important feature of the editing facility is the "Side A/B" editing button, which allows the user to specify the side of the tape on which each tune should be recorded. The result — far greater editing flexibility.

Auto edit mode

When the recording time and the auto edit mode are set, the tunes are automatically recorded in the same order as on the CD for side A. When side A of the tape is nearly full, if any of the subsequent tunes on the CD will fit into the remaining time, their tune numbers will be displayed and one of these can be specified to be recorded on side A. After this, the remaining tunes will be recorded on side B, until the tape reaches its end. But, if the total playing time of the remaining tunes exceeds the maximum recording time of side B, tune numbers are displayed in the same way, and the user can select which is to be recorded.

• Example of auto editing recording

— Side A = 1, 2, 3 ... 6 Side B = 7, 8, ... 12

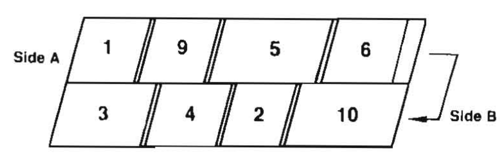


Program (manual) edit mode

With this manual editing function, the user can specify his/her own favorite tunes, so that their total length matches the recording time of the tape used. Using the programmed playback function of the CD player, the user can assign the tunes as desired, together with the side of the tape to be used for recording. After inputting the recording time of the tape to be used, each time you specify a tune, the remaining time will be counted down by the playing time of the tune so you can see at a glance when there's too little recording time left.

• Example of program editing recording

— Side A = 1, 9, 5, 6 .... Side B = 10, 2, 4, ...

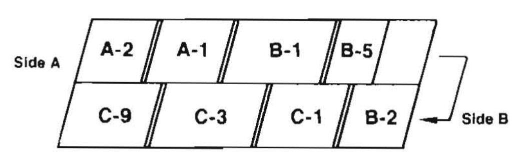


Multi-disc editing mode

With this multi-disc editing mode, the user can program tunes from a number of discs to be transferred to a single tape. The user can assign his or her favorite tunes, even if they are on different discs, in a single programming sequence. With our auto-changer models, operation is more convenient. Since six discs are loaded at one time, the user can assign the tunes to be transferred to tape without having to replace discs during the recording process.

• Example of multi-disc editing recording

— Side A = A-2, A-1, B-1, B-5, ... Side B = B-2, C-1, C-3, C-9, ...



3-way editing facility together with DDRP function

As the 3-way editing function is one of our most powerful features, you should emphasize how very convenient it is to potential customers who may want to tape their compact discs to create customized tapes. The descriptions below are how to demonstrate the procedures when the CD player and cassette deck are COMPU LINK components, but do not have the DDRP function. For the most effective demonstrations, you should use the XL-Z431BK CD player with the DDRP function together with a cassette deck which also has the DDRP function. If you do this, you can effectively demonstrate both the CD Editing features and the DDRP function. In this case, you can omit two procedures; 'set the cassette deck to the REC-PAUSE mode' and 'press (PLAY/PAUSE) of the CD player to start recording' — steps 4) and 5) in auto edit recording, steps 10) and 11) in program edit recording, and steps 8) and 9) in multi edit recording. This is how the DDRP function makes recording easier. We recommend you practice the procedure before performing an actual demonstration.

■ Auto edit recording

1) Load a disc and press (STOP/CLEAR).

STOP/CLEAR

2) Press (EDITING) once.

EDITING

3) Press (SIDE A/B).

SIDE A/B

If required, input the recording time according to the tape length with the numeric keys, before pressing (SIDE A/B).

4) Set the JVC COMPU LINK cassette deck to the REC-PAUSE mode.

REC PAUSE

5) Press (PLAY/PAUSE) on the CD player to start recording.

PLAY/PAUSE

With this, when the last tune programmed to be recorded on side A of the tape finishes, the deck and the CD player stop automatically, while the tunes programmed to be recorded on side B remain in memory. To continue recording, turn over the tape and press (PLAY/PAUSE) again.

■ Program edit recording

1) Load a disc and press (STOP/CLEAR).

STOP/CLEAR

2) Press the (EDITING) button twice.

EDITING

3) Press (SIDE A/B).

SIDE A/B

If required, input the recording time according to the tape length with the numeric keys before pressing (SIDE A/B).

4) Designate the tune with the numeric keys.

1 2

5) While the track number is blinking, press the (PRGM) button.

PRGM

6) Repeat steps 4 and 5 to program any other tunes to be recorded.

7) When only a few minutes are left for recording on Side A, the numbers of tunes which would fit into the remaining time will blink, so that you can select the tune to be recorded.

8) After programming is complete, press (SIDE A/B).

SIDE A/B

9) Repeat steps 4 — 6 for the other side of the tape

10) Set the cassette deck to the REC-Pause mode.

REC PAUSE

11) Press (PLAY/PAUSE) of the CD player to start recording.

PLAY/PAUSE

With this, when the last tune programmed to be recorded on side A of the tape finishes, the deck and the CD player stop automatically, while the tunes programmed to be recorded on side B remain in memory. To continue recording, turn over the tape and press (PLAY/PAUSE) again.

■ Multi edit recording

This is an exclusive JVC feature, so, if you don't have much time, this should be demonstrated rather than the other editing facilities.

1) Load a disc and press (STOP/CLEAR).

STOP/CLEAR

2) Press (EDITING) three times.

EDITING

3) Press (SIDE A/B).

SIDE A/B

If required, input the recording time according to the tape length with the numeric keys before pressing (SIDE A/B).

4) Designate the tune with the numeric keys.

1 2

5) While the track number is blinking, press the (PRGM) button.

PRGM

6) Repeat steps 4 and 5 to program any other tunes required to be recorded.

7) When only a few minutes are left on side A, the numbers of tunes which would fit in the remaining time will blink, so that you can select the tune to be recorded.

8) After programming tunes is complete, set the cassette deck to the REC-PAUSE mode.

REC PAUSE

9) Press the (PLAY/PAUSE) to start recording.

PLAY/PAUSE

When all the required tunes have been recorded from the first disc, the deck and the CD player stop automatically. Replace the disc to continue recording. Even when the OPEN/CLOSE is pressed to replace the disc, the remaining time is still held in memory and you can continue recording continuously, replacing discs one after another, as long as there is room on the tape.

Feature Highlights of '90 Cassette Decks

1

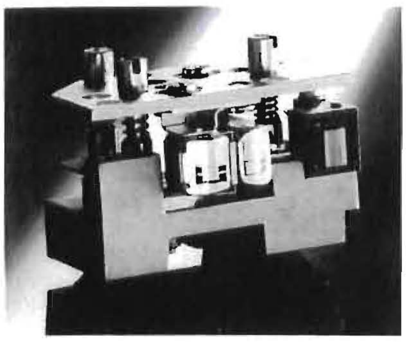
Discrete three-head cassette decks with "fine amorphous" heads and closed-loop dual-capstan drives
(TD-V1010TN)

2

DDRP *(Dynamics Detection Recording Processor)*; ***easy recording from CD players to cassette decks with a wider dynamic range***

3

Dolby HX-PRO is provided in all models *(except for TD-W203BK and TD-W103BK)* ***to improve high-frequency characteristics***



"Fine amorphous" heads with extended high-frequency response; closed-loop dual-capstan mechanism for optimum tape tension at all times

The recording and playback heads are both made from amorphous ferrite. These heads combine an improved magnetic performance and excellent anti-abrasion characteristics with an extended high-frequency response. The closed-loop dual-capstan mechanism uses a highly stable direct-drive motor for optimum head-to-tape contact and reduced wow and flutter.

One-touch recording system sets the optimum recording level

This year's new cassette decks (except for the TD-W103BK) are all provided with the DDRP function and the TD-R431BK and TD-X331BK are provided with a DDRP switch. When the DDRP switch on the CD player (XL-Z431BK) is pressed, "peak search" is performed, scanning the disc and determining the optimum recording level. After the "peak search" has been completed, the cassette deck sets the input level according to the type of tape and enters the recording standby mode. Then, with a cue from the CD player, the cassette deck enters the recording mode to record with the widest possible dynamic range. In this way, the complicated operations required to set the recording level manually are eliminated.

Dolby HX-PRO headroom extension circuit expands the effective dynamic range of tape

Dolby HX Pro developed by Dolby Laboratories controls the bias current so that the effective bias is constant, even when there are fluctuations in the high-frequency components of the input signal. This greatly improves the high-frequency saturation level while reducing low-frequency signal level variations and distortion. In combination with Dolby B C noise reduction system, it makes possible recording with a dynamic range equivalent to that of a digital source program.

Feature Comparison Chart

Cassette Decks
(Single-Transport Models)

		TD-V1010TN	TD-V711BK	TD-V531BK	TD-R431BK	TD-X331BK
Mechanism						
3-Head Configuration		✓	✓	✓		
Fine-Amorphous Head		✓				
Full-Logic Control		✓ (Silent)	✓ (Silent)	✓ (Silent)	✓	✓
Closed-Loop Dual Capstan		✓	✓	✓		
Direct Driven Motor		✓	✓			
Hi-Fi U-Turn Auto-Reverse					✓	
Cassette Shell Stabilizer		✓		✓		
Circuit						
Dolby HX Pro		✓	✓	✓	✓	✓
Dolby B/C Noise Reduction		B/C × 2	B/C × 2	B/C × 2	B/C	B/C
Bias Adjust		✓	✓	✓	✓	✓
Level Calibration		✓				
CD Direct Inputs		✓	✓	✓		
Direct Inputs		✓	✓			
Line Inputs		✓	✓	✓	✓	✓
3-Block Construction (Separated)		✓				
Display						
Display Panel		FL	FL	FL	FL	LED
Digital Peak Display		✓	✓			
Digital Counter		✓	✓	✓	✓	
Operation Mode Indicator		✓	✓	✓	✓	✓
Display ON/OFF		✓				
Function						
DDRP				✓	✓ (Switch)	✓ (Switch)
Auto Monitor		✓	✓	✓		
Monitor Switch		✓	✓	✓		
Peak Call Switch		✓	✓			
MPX Filter Switch		✓	✓	✓		
Auto Tape Selector		✓	✓	✓	✓	✓
Timer Start (Rec/Play)		✓	✓	✓		
Auto Rec Mute		✓	✓	✓	✓	✓
Music Scan		✓	✓	✓		
Input Balance		✓	✓	✓	✓	✓
Direction Switch					✓	
Remote Control (Provided)		✓				
Others						
Gold-Plated Terminals (In/Out)		✓	✓	✓		
Headphone Output	with Volume	✓	✓			
	Fixed			✓	✓	✓
COMPU LINK Component		✓	✓	✓	✓	✓

Feature Comparison Chart
Cassette Decks
(Double-Transport Models)

		TD-W901BK	TD-W803BK	TD-W503BK	TD-W303BK	TD-W203BK	TD-W103BK
Mechanism							
Hi-Fi U-Turn Auto-Reverse		✓ (A/B) (Quick)	✓ (A/B)	✓ (A/B)	✓ (B)		
Record/Playback		A/B	A/B	B	B	B	B
Full-Logic Control		✓	✓	✓	✓	✓	
Display							
Display Panel		FL	LED	LED	LED	LED	LED
Digital Counter (Twin)		✓					
Operation Mode Indicator		A/B	A/B	A/B	B	B	
Direction Indicator		A/B	A/B	A/B	B		
Circuit							
Dolby HX Pro		✓	✓	✓	✓		
Dolby Noise Reduction		B/C	B/C	B/C	B/C	B/C	B
Pitch Control				✓			
Input Balance		✓	✓	✓			
Mic Inputs (L/R)				✓			
Timer Start (Rec/Play)		✓	✓				
Headphone Output (Fixed)		✓	✓	✓	✓	✓	
Function							
DDRP			✓	✓	✓	✓	
Hi-Speed Dubbing		✓	✓	✓	✓	✓	✓
Synchro Dubbing		✓	✓	✓	✓	✓	✓
Continuous Play		✓	✓	✓	✓	✓	✓
Music Scan	Multi (A/B)	✓					
	Single (Deck A/B)		✓	✓			
Synchro Rec Mute		✓	✓	✓	✓	✓	
Auto Rec Mute		✓	✓	✓	✓	✓	
Auto Tape Selector (Deck A/B)		✓	✓	✓	✓	✓	✓ (Deck A)
Remote Control (Provided)		✓					
Flip-down Door		✓					
COMPU LINK Component		✓	✓	✓	✓	✓	

Lineup of '90 Cassette Decks

Single-Transport Models

Discrete 3-head cassette deck with "Fine" amorphous heads and remote control

Titanium-finished

TD-V1010TN

Discrete 3-Head Cassette Deck

SUPER DIGIFINE COMPU LINK Component

- Discrete 3-head configuration with "Fine Amorphous" heads
- 2 "Direct" inputs
- "Silent mechanism" head drive assembly
- Low resonance design
- Closed-loop dual-capstan drive with direct-drive motors
- PCOCC★coil/lead wiring and OFC☆-plated circuit boards

★PCOCC Perfect Crystal by Ohno Continuous Casting
☆OFC Oxygen-Free Copper

Discrete 3-head cassette deck with two direct inputs

TD-V711BK

SUPER DIGIFINE COMPU LINK Component

Discrete 3-Head Cassette Deck

- Discrete 3-head configuration
- Closed-loop dual-capstan drive with direct-drive motors
- 2 "Direct" inputs
- "Silent mechanism" head drive assembly
- PCOCC★ coil/lead wiring and OFC☆-plated circuit boards

NEW *High cost/performance 3-head deck with closed-loop dual-capstan drive and DDRP*

TD-V531BK

DIGIFINE COMPU LINK Component

Combination 3-Head Cassette Deck

- 3-head configuration
- New DDRP
- Dolby HX-Pro
- Closed-loop dual-capstan drive
- "Silent mechanism" head drive assembly
- "Direct" input

NEW *Hi-Fi U-Turn quick auto-reverse deck with DDRP and Dolby HX-Pro*

TD-R431BK

COMPU LINK Component

Hi-Fi U-Turn Quick Auto-Reverse Cassette Deck

- Hi-Fi U-Turn quick auto-reverse with Flip Reverse Heads
- New DDRP
- Dolby HX-Pro
- Bias adjustment
- Dolby B/C noise reduction

NEW Full-logic control cassette deck with DDRP and Dolby HX-Pro

- Computer-controlled full-logic control
- New DDRP
- Dolby HX-Pro
- Dolby B/C noise reduction
- Bias adjustment

TD-X331BK COMPU LINK Component

Full-Logic Control Cassette Deck

Double-Transport Models

Dual record/play Hi-Fi U-Turn quick auto-reverse double cassette deck with remote control

- Twin Hi-Fi U-Turn quick auto-reverse with Flip Reverse Heads
- Dual record/playback tape transports
- Twin Dolby HX-Pro
- Computer-controlled full-logic control
- Sequential or simultaneous two-tape recording

TD-W901BK DIGIFINE COMPU LINK Component

Hi-Fi U-Turn Quick Auto-Reverse Double-Mechanism Cassette Deck

NEW Dual record/play Hi-Fi U-Turn auto-reverse double cassette deck with DDRP and HX-Pro

- Twin Hi-Fi U-Turn auto-reverse with Flip Reverse Heads
- New DDRP
- Dual record/playback transports
- Dolby HX-Pro
- Sequential or simultaneous two-tape recording

TD-W803BK DIGIFINE COMPU LINK Component

Hi-Fi U-Turn Auto-Reverse Double-Mechanism Cassette Deck

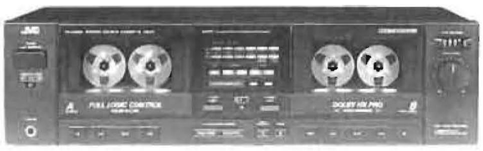
NEW Dual Hi-Fi U-Turn auto-reverse double cassette deck with pitch control, microphone inputs and DDRP

- Twin Hi-Fi U-Turn auto-reverse with Flip Reverse Heads
- New DDRP
- Record/playback and play-only tape transports
- Dolby HX-Pro
- Computer-controlled full-logic control

TD-W503BK COMPU LINK Component

Hi-Fi U-Turn Auto-Reverse Double-Mechanism Cassette Deck

NEW Full-logic control double cassette deck with Hi-Fi U-Turn auto-reverse mechanism and DDRP



- Hi-Fi U-Turn auto-reverse with Flip Reverse Heads
- New DDRP
- Record/playback and play-only tape transports
- Dolby HX-Pro
- Computer-controlled full-logic control

TD-W303BK

COMPU LINK Component

Full-Logic Control Double-Mechanism Cassette Deck with Hi-Fi U-Turn Auto-Reverse

NEW Full-logic control double cassette deck with DDRP

- Computer-controlled full-logic control
- Record/playback and play-only tape transports
- New DDRP
- Dolby B/C noise reduction
- Continuous play of two tapes

TD-W203BK

COMPU LINK Component

Full-Logic Control Double-Mechanism Cassette Deck

NEW Basic double cassette deck with Dolby B noise reduction

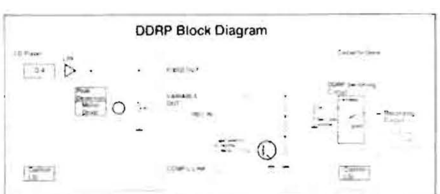
- Record/playback and play-only tape transports
- Dolby B noise reduction
- High-speed editing with synchro start
- Continuous play of two tapes
- Auto tape selector (Deck A)

TD-W103BK

Double-Mechanism Cassette Deck

Feature reference

Technology/Function | **Benefits**



DDRP (Dynamics Detection Recording Processor) function

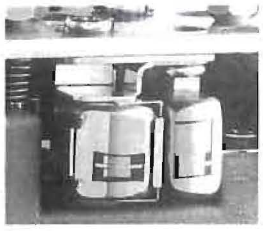
(TD-V531BK, TD-R431BK, TD-X331BK, TD-W803BK, TD-W503BK, TD-W303BK, TD-W203BK)

This function allows automatic recording from a CD player with the optimum recording level and widest dynamic range by using the COMPU LINK Control System. When a button on the CD player is pressed, the CD player performs "peak search" by scanning the compact disc at high speed to determine the peak signal level so that the recording level can be optimized; after the "peak search" has been completed, the cassette deck enters the recording standby mode and detects the type of tape to select the best input level. When the optimum recording level has been determined, the cassette deck starts recording. (In case of the TD-R431BK and TD-X331BK, as a DDRP switch is provided, after pressing this button, the same procedures are performed automatically.)

Normally, when recording a compact disc onto a cassette tape, adjusting the recording level can take a long time. However, with this system, the level adjustment is performed by simply pressing a button on the CD player and high-quality recording is performed automatically and easily with the widest possible dynamic range.

Technology/Function	Benefits

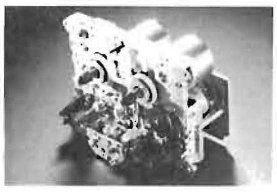
Discrete 3-head configuration (TD-V1010TN, TD-V711BK)



The TD-V1010TN incorporates three heads with fine amorphous ferrite heads for recording and playback while the TD-V711BK incorporates three heads with an SA head for recording and an amorphous ferrite head for playback. This means the azimuth, tilt, and tracking can be aligned head by individual head for the best results, without any compromise.

The discrete 3-head design allows the user to monitor the off-tape sound during recording. This means the user can directly check and compare the recorded sound with that from the original source. Especially when recording with Dolby* noise reduction which requires precise calibration, the user can check the Dolby-decoded playback signal and verify optimum results because double Dolby circuits are provided, for both recording and playback.

Closed-loop dual capstan mechanism
(TD-V1010TN, TD-V711BK, TD-V531BK)



A high-precision "closed-loop dual capstan" mechanism is used in the TD-V1010TN, TD-V711BK and TD-V531BK to keep the head-to-tape contact stable and reduce wow and flutter.

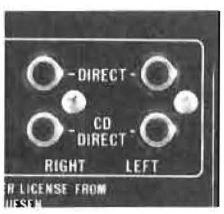
With this mechanism, there are capstans and pinch rollers on both side of the independent heads to keep the tape tension constant, from beginning to end of tape, ensuring stable head-to-tape contact.

Dolby HX Pro (All models except TD-W203BK and TD-W103BK)

This is another circuit developed by Dolby Laboratories, but it is for expansion of tape's effective dynamic range rather than noise reduction. The Dolby HX Pro headroom extension circuit was developed to compensates for an inherent weakness in tape recording systems.

Dolby HX Pro controls the bias current so that the effective bias is constant, even when there are fluctuations in the high-frequency components of the input signal. This greatly improves the high-frequency saturation level while reducing low-frequency signal level variations and distortion.

Direct input jacks (TD-V1010TN, TD-V711BK, TD-V531BK)



Models TD-V1010TN, TD-V711BK and TD-V531BK are equipped with CD DIRECT and/or DIRECT IN jacks; these jacks can be directly connected to a CD player or other source components without passing through the amplifier.
Further, these models have the shortest possible signal transmission path.

When a source signal is directly input to the cassette deck without passing through the amplifier, the recording can be performed without signal loss, so that playback sound is far superior to that from a cassette deck without DIRECT jacks. Moreover, because the signal transmission lines are shorter, the sound is improved because unnecessary circuitry does not introduce noise or degrade the deck's characteristics.

Refer to "Technical Notes" page 50 for more information.

"Fine amorphous" heads (TD-V1010TN)

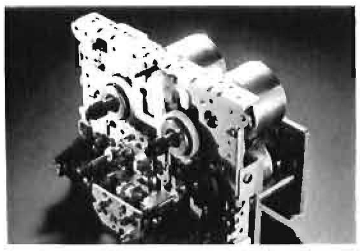
Based on our SA (Sen-Alloy) heads, with their higher magnetic performance, technically "higher maximum flux density", and excellent anti-abrasion characteristics, a newly developed "fine amorphous" material with an extended high-frequency response is used in the heads of our top-end "quality-oriented" model.

Although the amorphous material is basically unstable and therefore has a shorter service life, our improved "fine amorphous" material features well-balanced characteristics, to combine an extended high-frequency response and a higher magnetic flux density with a longer service life as SA heads. It makes possible extremely pure signal transmission, while the wider dynamic range and flat frequency response of digital sound is maintained, for music of the highest quality.

Hi-Fi U-Turn auto-reverse with Flip Reverse Head
(TD-R431BK, TD-W901BK, TD-W803BK, TD-W503BK, TD-W303BK)

JVC's Hi-Fi U-Turn auto-reverse models incorporate a high-precision Flip Reverse Head system which rotates through 180 degrees extremely quickly and accurately when the tape running direction is switched.

This enables the user to change the tape running direction instantly during playback or recording. The user can easily listen to tapes or make recordings for long periods, with extremely short interruptions, while maintaining the same superb audio performance in both the forward and reverse directions.
Further, with the TD-R431BK and TD-W901BK, you can enjoy unbroken music without any time lag as the tape is reversed immediately at the leader tape. (Quick Reverse)



Two-motor full-logic mechanism (TD-V1010TN, TD-V711BK, TD-V531BK)

JVC's two-motor full-logic control models employs two-motors for cassette operation; one is used exclusively for driving the capstan (one of the most important parts in any magnetic recording machine) and the other for driving the reels. These two motors are controlled by sophisticated micro-computer.

The capstan motor ensures that the tape runs at precisely the correct speed, while the reel motor drives the supply and take-up reels smoothly, in the rewind and fast-forward modes as well as in normal playback. This system allows users to operate the decks easily with "feather-touch" control that provides quick response and the easiest possible operation, while maintaining precise control over tape movement.

"Silent" head assembly drive (TD-V1010TN, TD-V711BK, TD-V531BK)

Our "silent mechanism" models provide an additional motor exclusively to drive the mechanism which includes the head assembly, for quick and accurate response as well as rigid construction.

The head assembly which is enclosed in a rigid structure can be controlled quickly and accurately in response to the user's operations.

Cassette shell stabilizer (TD-V1010TN, TD-V531BK)

To reduce acoustic modulation noise which occurs due to external vibrations, sound fed back from the speakers, etc., which could affect sound quality, the mechanism was designed employing a number of anti-resonance techniques. For example, the chassis supporting the tape drive mechanism is made from rigid die-cast aluminum, while the front panel is molded from non-resonating high-density resin. For even greater stability, a cassette stabilizing pad is used to hold the cassette firmly by pressing against the center of the shell, in addition to the conventional cassette holder mechanism.

While the tape is running during recording and playback, if the cassette shell is subject to vibrations or resonates in any way, small fluctuations may occur in the recorded/playback signal, which would result in degraded sound quality. The cassette shell stabilizer, as you will realize from its name, effectively stabilizes the cassette by dampening any vibrations and resonance so they can never affect signal quality. Together with our other anti-vibration, non-resonance techniques, this acts to virtually eliminate acoustic modulation noise.

Refer to "Technical Notes" page 50 for more information.

PCOCC and LC-OFC head winding and circuit board plating (TD-V1010TN, TD-V711BK)

PCOCC stands for Perfect Crystal by Ohno Continuous Casting process. This process produces high-purity copper which is perfectly crystalline, for improved electron flow. LC-OFC means Linear Crystal Oxygen-Free Copper. With less oxygen, the wires transmit the signal more efficiently, and with less coloration. JVC uses PCOCC and LC-OFC of the highest purity for the coils in head cores, for signal leads, and for plating on the main circuit boards.

This results in superior transmission characteristics, effectively reducing the transmission loss and degradation of the signals. The signals recorded and reproduced by these decks are clearer and more musically accurate.

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Separate circuit construction (TD-V1010TN, TD-V711BK)

The recording amplifier section, playback amplifier section, signal control & display section, and power supply section are all independently mounted on separate PC boards.

Since these decks are constructed with independent blocks, both electrical and magnetic interference are totally eliminated.

DC-configured amps and low-impedance voltage-tracking regulated power supply (TD-V1010TN, TD-V711BK)

In these models, only specially selected components and circuits are used, including DC-configured amps which ensure high linearity, voltage-tracking power supplies using an independent reference voltage generator rather than a simple diode to keep the ground potential accurately at zero, and constant-voltage circuits to reduce the output impedance to 1/10 of the usual value. The result is an incredibly flat frequency response from 20 Hz to beyond 16 kHz.

Constant fluctuations in the voltage supplied from the household AC outlet cause changes in the voltage on the secondary side of the recorder's transformer, which are then passed on the deck's amplifying circuits. This results in inaccurately amplified signals and degraded sound. These features effectively prevent this phenomenon, working together and assuring better linearity and lower distortion as well as the lowest possible noise. They provide an excellent response throughout the audio spectrum. Therefore, the user can expect impeccable audio response even when recording from a digital source, without coloring and tone deterioration.

Recording calibration control (Bias/Level)

(All single-transport models)

All single-transport models are equipped with a recording calibration control which allows fine adjustment of the bias current added to the music signal before recording. Moreover, in our top-of-the-line TD-V1010TN, an independent level alignment control is provided in addition to the bias adjustment control.

Actually, depending on the tape used, there are minute difference in level and the amount of bias required for optimum recording. Bias adjustment compensates the amount of bias applied to signal being recorded. With this, the user can compensate the characteristics of the tape itself. For instance, decreasing the amount of bias — possible with certain tapes — will result in extended high-frequency response, and increasing it will have the reverse effect. Therefore, the user can obtained the best possible recording with the tape he or she is using. With the TD-V1010TN, since an independent level adjustment control is also provided, enabling sensitivity calibration within a range of ±3 dB, even more accurate, further optimized tape response can be obtained.

Dolby* B/C noise reduction systems (All models)**

Dolby noise reduction systems effectively reduces the noise in recording, by approx. 10 dB with B-type noise reduction and by approx. 20 dB with C-type noise reduction; the effect is especially noticeable in the high frequency range.

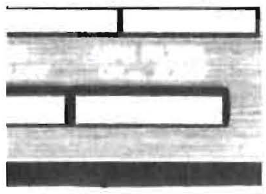
Noise reduction provides superior high frequency response while background tape hiss (inevitable in tape recording) is reduced so as to be almost inaudible. The result is clear and brilliant high frequencies with low noise, the biggest benefit from recent advances in digital audio technology. Dolby B noise reduction is provided so that the large number of music tapes encoded with Dolby B noise reduction and which the user probably has in his or her collection can be played back with full noise reduction. With its greater effectiveness at all frequencies, most users will prefer to use Dolby C noise reduction for new additions to their libraries.

* DOLBY and the double D symbol are trademarks of Dolby Laboratories Licensing Corporation.
** The TD-W103BK is provided with only Dolby B noise reduction system.

Double Dolby circuits (TD-V1010TN, TD-V711BK, TD-V531BK)

In the TD-V1010TN, TD-V711BK and TD-V531BK, independent Dolby noise reduction systems are provided in the recording and playback circuits and these can work simultaneously, even during recording, to derive optimum results from the 3-head construction.

Since these decks have independent heads for recording and playback, off-the-tape sound (the sound which has just been recorded by the record head) can be monitored (using the playback head) even when the Dolby circuit is used for recording. The user can check the effectiveness of recording through the Dolby B or C circuits and confirm the improvement that it provides; he or she will be able to hear that the source sound (even from compact discs) is being recorded as it is, with maximum fidelity.



MPX filter ON/OFF switch (TD-V1010TN, TD-V711BK, TD-V531BK)

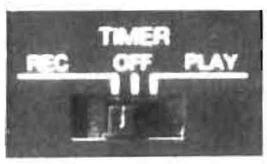
These models are provided with an MPX filter for more effective recording of FM broadcasts from a tuner, while using the Dolby NR system.

With the MPX filter switch ON, the misoperation of the Dolby NR circuit is avoided by filtering out the 19 kHz FM pilot signal. Further, if the tuner does not have a MPX filter or if its filter is inadequate, using the MPX filter switch of the deck avoids sound quality deterioration due to the malfunctioning of the Dolby NR circuit.

Auto tape selector (All models)

All recent tape cassettes have identification holes in the edge for the detection of Normal, Chrome (CrO_2) and Metal tape. When the cassette is loaded, the optimum EQ/bias characteristics will be set automatically by detecting these identification holes.

The user can operate the deck immediately after loading a cassette. It is not necessary to check and reset a switch to correspond to the type of tape used for playback or recording.



Timer start (record/playback)
(TD-V1010TN, TD-V711BK, TD-V531BK, TD-W901BK, TD-W803BK)

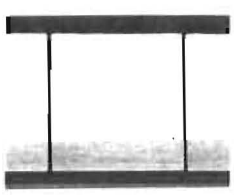
When used together with an optional audio timer, the tape transport can be started at any preset time for either playback or recording. While the deck is in the timer standby mode (before and after playback/recording), the capstan and tape are not compressed between the pinch roller and capstan. This means that the tape and the rubber pinch roller are not subject to physical distortion, even if timer operation is performed repeatedly.

The deck can also be used for unattended recording or as an alarm in the morning. The user can wake up to his or her favorite music, or important programs can be recorded, even when the user is in bed or not at home. Since a computer-controlled mechanism is used, timer recording can be performed repeatedly up to the end of tape without any worry about worn heads, pinch rollers and tape.

Synchro Rec Mute (All dual-transport models except TD-W103BK)

In our dual-transport models, a Synchro Rec Mute function is provided for added convenience.
When editing from one deck to another, if the Music Scan is activated for the playback deck, the recording deck will automatically perform Rec Mute operation.

Synchro Rec Mute allows the recording deck to enter the record-pause mode after leaving a 4-second interval automatically when the playback deck enters the Music Scan mode during dubbing. This feature permits the user to make his/her own customized tapes easily, with only their favorite tunes copied from longer tapes



Auto Rec Mute (All models except TD-W103BK)

By simply touching a button, a non-recorded section of approx. 4 seconds will be left automatically, with the deck set to the record pause mode.

Auto Rec Mute instructs the recording deck to leave a blank section of approx. 4 seconds automatically, with the user just pressing button It is designed for use when there's a narration or commercial in an FM broadcast that's being recorded, etc. The user can easily leave blank sections during recording for uniform gaps between one song and the next. These uniform gaps between tunes allow various scanning operations to work correctly.

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Music Scan (TD-V1010TN, TD-V711BK, TD-V531BK, TD-W901BK, TD-W803BK, TD-W503BK)

By pressing FF or REW together with PLAY, the deck enters the fast-forward or rewind mode to skip to the next or previous tune by detecting the blank sections between tunes.

With tape recordings, finding the beginning of a tune is difficult because you can't see the gaps between tunes. However, with JVC's Music Scan function, the user can skip to the start of the next or previous tune quickly using this automatic function

Synchro Start Dubbing (All dual-transport models)

Our top-end dual-transport decks the TD-W901BK and TD-W803BK incorporate high-quality dual record/playback tape transports while other dual-transport decks incorporate record/playback and play-only tape transports. They provide a level of convenience which is impossible with conventional signal-transport decks. One benefit is the tape-to-tape/editing capability. For this purpose, the Synchro Start function lets you set one transport to the playback mode and the other to the recording modes simultaneously, by pressing a single button.

Double-transport design allows the user to dub or edit tapes very easily, by simply pressing the Synchro Dubbing button (except for the TD-W103BK). The entire contents of one tape can be directly transcribed onto another tape. The user no longer has to perform complicated operations with two decks, for problem-free editing.

Hi-speed editing (All dual-transport models)

All JVC dual-transport models feature a hi-speed editing function that permits copying at twice the normal tape running speed in the synchro editing mode.

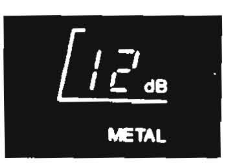
While the Synchro Dubbing button (Synchro Start button for the TD-W103BK) allows the playback and recording decks to start simultaneously, the High-Speed button enables the same operation to be performed at twice normal speed, cutting the editing time in half. For example, the user can duplicate both sides of a C-60 tape in only 30 minutes. Even when the tape is running at double speed with frequencies that are double those at normal speed, all JVC models are able to maintain full audio performance with their hard and durable heads. Even the highest frequency sound will not deteriorate in duplication. Of course, JVC decks with the High-Speed mode also have a Normal Speed editing mode, so the user can listen to tapes while editing them

Continuous playback (All dual-transport models)

Another feature provided in all JVC's dual-transport decks is continuous playback. Deck B automatically starts playback when the end of tape is reached in Deck A, after which the first tape is automatically rewound (in the case of non auto-reverse decks), to be played again. In the case of the TD-W103BK, Deck B starts playback when playback by Deck A ends; when playback of the tape in Deck B ends, Deck B enters the stop mode.

* In case of non auto-reverse decks.

The continuous playback mode allows endless playback over long periods; while one tape is being played, the other is automatically rewound to the start so it can be played again. Two tapes loaded in Decks A and B can be played back-to-back continuously, for playback without a break. The user can enjoy the two tapes over and over again continuously without touching the deck, until the tapes are removed from the deck. This feature is very convenient when you want background music, etc.



Digital peak display (TD-V1010TN, TD-V711BK)

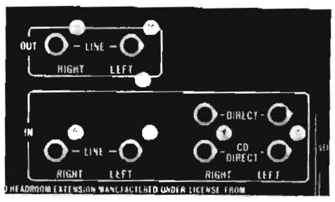
Digital Peak Display shows momentary transient input or output levels digitally, in increments of 1 dB. The highest peak level can be recalled at any time as it is stored in the built-in microcomputer.

This easy-to-see indication permits the user to more accurately adjust the critical recording level and obtain optimum dynamic range, ensuring the best recording results, even with digital sources.

Electronic digital counter
(TD-V1010TN, TD-V711BK, TD-V531BK, TD-R431BK, TD-W901BK)

An easy-to-see 4-digit electronic digital counter is incorporated in these models to be used for a variety of functions — as tape counter, a tape remaining time display, in music scanning, etc. — depending on the cassette deck.

The TD-V1010TN has two-mode electronic digital counter and can be used as a tape counter and as a tape remaining time display; it can also be switched off when not required, for reduced interference. Further, the TD-W901BK has dual electronic digital counters so that the movement of the tapes in each of the two mechanisms can be checked at a glance



Gold-plated terminals and quality parts
(TD-V1010TN, TD-V711BK, TD-V531BK)

In these models, the input/output terminals are plated with gold and stacked film capacitors are used instead of coiled ones to maintain their high quality audio performance for a longer period of time.

Gold-plate terminals have better electrical contact and resistance to corrosion so that distortion is minimized even after years of use. The initial superb audio performance will not be degraded, even after years of use, and the user will never have to worry about aging causing deterioration of the recording/playback characteristics. The stacked film capacitors have a lower inductance than that obtained with coiled capacitors, resulting in a better high-frequency response and greater resistance to vibrations.

Variable pitch control (TD-W503BK)

The TD-W503BK has a unique pitch control for Deck A, which permits the user to vary the tape speed in playback. By changing the tape speed, the pitch (frequency) of the recorded sound can be changed when playing back tapes.

Since the pitch of the playback sound can be varied, any tune can be played back with a pitch that meets the user's requirements. With a microphone or electronic musical instrument plugged into the stereo microphone inputs of the TD-W503BK, the user can record his or her own voice or instrumental backing over an accompaniment played back in deck A using deck B, with perfect pitch.

Fluorescent/LED multi-peak indicator
(FL: TD-V1010TN, TD-V711BK, TD-V531BK, TD-R431BK, TD-W901BK; LED: TD-X331BK, TD-W803BK, TD-W503BK, TD-W303BK, TD-W203BK, TD-W103BK)

A pair of fluorescent peak level meters or a pair of 6-LED indicators is used to show the moment-to-moment peak of the signal input to the deck during recording. It also shows the signal level from the tape during playback.

The user can check the peak transients in the recording signal. The optimum recording level can be set for each type of tape, and recordings can be made with widest possible dynamic range.

High bias frequency (TD-V1010TN, TD-V531BK)

In the TD-V1010TN, the bias frequency in recording is set at an extremely high level — 210 kHz — one of the highest levels used in 3-head models. This makes possible the great improvement of the overall characteristics of the TD-V1010TN.

In recording, high-frequency current is added to the music signals being recorded; without it, recording would not be possible. This is called the "bias" current, and frequencies of over 100 kHz are used in most decks. The TD-V1010TN uses an even higher frequency bias current, so that beats which could occur due to resonance with the frequency of the music signal are minimized, and the resultant sound is clearer than ever.

For Further Purer and Direct Signal Transmission — Another improvement in cassette deck technology

■ Why is "acoustic modulation noise" a problem?

As we enter the age of digital audio, cassette decks have reached maturity and it would seem that there is no way to improve their specifications.

Although it's not reflected in their specifications, there are differences between the components that make up a stereo system; acoustic resonance, external vibrations, etc. still affect sound quality.

Through our long experience and detailed research, we found that this was also true with cassette decks. As with turntables and CD players, acoustic resonance and external vibrations, due to the constantly changing sound pressure from the speakers, affect the tape drive mechanism, etc. and thus affect the quality of the signal being recorded. Although the effect is small, we feel we should do everything possible to eliminate it.

Changes in sound pressure and vibrations have an unfavorable effect on the mechanism of the cassette decks, most importantly on the tape transport mechanism, and result in a slight deterioration in sound quality.

This is called "Acoustic Modulation Noise".

Conventionally, "modulation noise" is measured by recording and playing back a 10 kHz signal with the cassette deck being tested. The peak represents the input signal. The sharper the peak and the smoother the slopes on either side, the higher the clarity of the sound. With the "acoustic modulation noise" test, however, the same measurement is conducted with the tape deck subject to 100-phon acoustic sound pressure generated by playing pink noise through speakers.

The "acoustic modulation noise" response curve shows how the sound is affected in an in-use situation.

We detected this "acoustic modulation noise" in the following way:

Fig. 1 shows the modulation noise characteristics of a closed-loop dual-capstan mechanism, with a 10 kHz input signal. The quicker and smoother the roll-off on both sides of the peak, the clearer the sound.

But when this mechanism is mounted in a cassette deck in which no special anti-vibration countermeasures have been applied, and measured in a stereo system, exposed to a sound field created by speakers, the response curve shown in Fig. 2 was obtained.

When the unit is placed in a sound field where pink noise at a 100-phon sound pressure is generated from speakers, the modulation noise characteristics were greatly degraded which indicates that the resultant sound quality would be degraded.

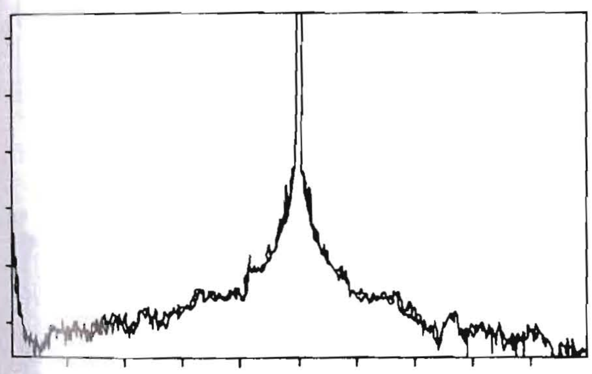


Fig. 1 Response of rigidly built deck

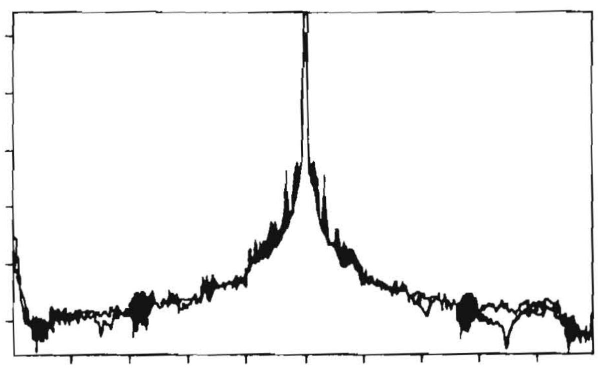


Fig. 2 Response of conventional deck

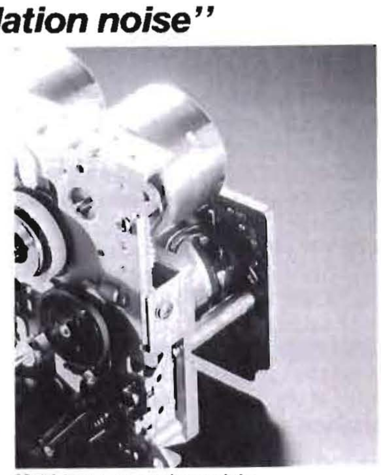
■ Countermeasures to prevent "acoustic modulation noise"

By analyzing "acoustic modulation noise", we found various countermeasures that could be used to reduce it.

To suppress acoustic modulation noise, first of all, the anti-vibration characteristics of the deck's tape transport mechanism were improved together with those of the front panel section which supports the mechanism, and also the entire unit, as well as suppressing vibrations of the cassette shell.

For this, our top model, the TD-V1010TN, uses a rigid diecast aluminum base to support the tape drive mechanism so that the tape passes the heads at precisely the correct speed at all times, while the PC board is supported by a thick multi-layer copper plate.

Diecast aluminum base of tape drive mechanism



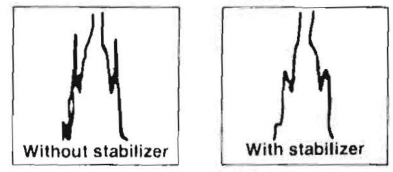
Multi-layer copper base plate

At the same time, a cassette shell stabilizer is provided to hold the cassette shell firmly in position; two stabilizing pads press against the center of the cassette shell, holding it firmly from both sides, to prevent minute oscillations of the tape as it is running due to vibrations, etc.

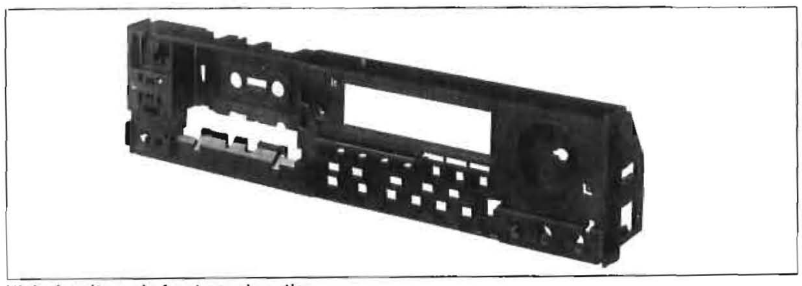
The front panel section is made of solid, high density resin with a rigidity 1.7-times that of normal resin and 1.5-times greater density, and the entire chassis is securely supported by the heavy, solid base using large insulators instead of the normal feet, so that the deck as a whole is protected from external vibrations.

By taking these countermeasures, acoustic modulation noise is greatly reduced and the response curve shown in Fig. 3 is obtained, demonstrating that the TD-V1010TN can reproduce sound with outstanding clarity when used in a stereo system, in actual in-use conditions.

Cassette shell stabilizer



Improvement in modulation cassette stabilizer



High-density resin front panel section

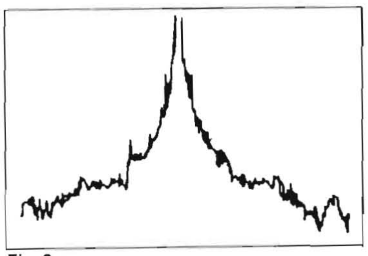
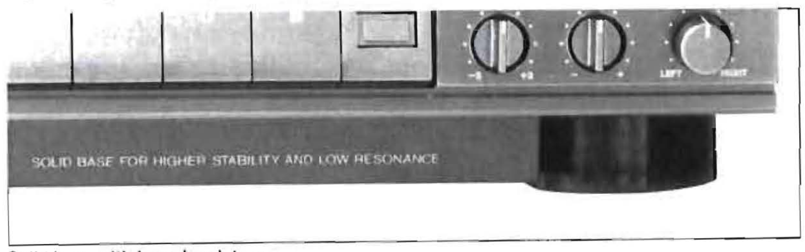


Fig. 3



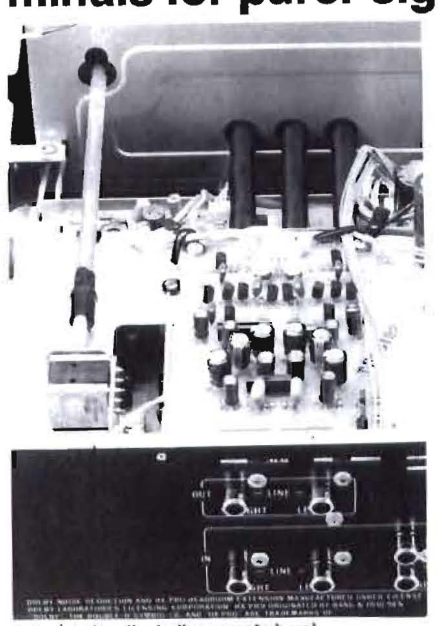
Solid base with large insulators

"DIRECT" Input Terminals for purer signal transfer

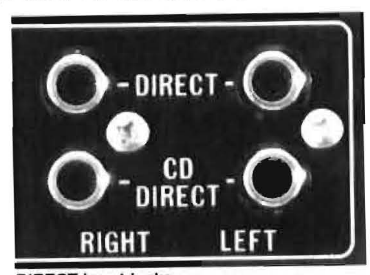
In addition to the regular analog input/output jacks, the TD-V1010TN, TD-V711BK and TD-V531BK provide the two pairs of "DIRECT" input jacks, enabling direct connection to a CD player or other source component.

Since the signal from a CD player (for example) does not pass through the amplifier, there is no possibility of it being degraded as would happen with the normal connection method. And because there is no source selector or REC selector between the source component and the cassette deck, there is no switching noise or any deterioration due to electrical contact; the high quality of sound from the digital source is transferred directly to the cassette tape.

Together with the shortest possible internal connection of our cassette decks using "remote bars", the total length of the circuit from the pickup of the CD player to the recording head of the cassette deck is made as short as possible. This is the real meaning of our "pure & direct" signal transmission concept.



Internal view (including remote bars)



DIRECT input jacks

AMPLIFIERS

Feature Highlights of '90 Amplifiers

1

The "Digital Pure-A Type II" — An even more advanced amplifier design, combining high power with class-A operation

DIGITAL PURE A

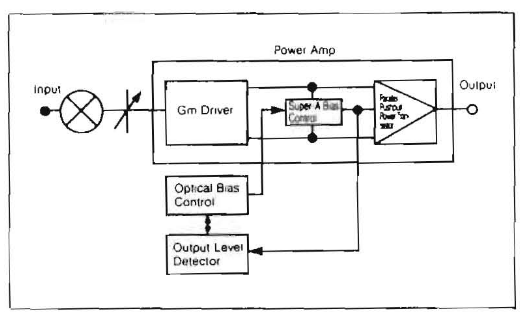
2

The "K2 Interface" completely isolates the digital and analog circuits

K2 INTERFACE

3

Our "Opt Super-A" circuitry is another milestone in amplifier design, for high-power and class-A operation with analog inputs

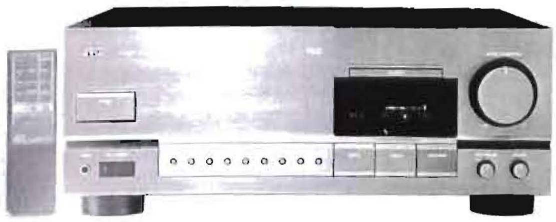


Feature Comparison Chart
Amplifiers

		AX-Z1010TN	AX-Z911BK
Power Output			
8 ohms (20–20 kHz) (watts per channel)		100 W	100 W
2 ohms (Dynamic Power)		320 W	
Digital			
K2 Interface		✓	
Digital Pure-A (Type II)		✓	
Digital Pure-A			✓
Digital Filter		✓ (8fs)	✓ (4fs)
DA Converter	18-bit Combination 4 DAC	✓	✓ (Dual 16-bit)
DAC Direct		✓	✓
Circuit			
Opt Super-A		✓	
2-Amp Construction		✓	
Remote Control (Provided)		✓	✓
Source			
Digital	Digital 1 (Optical)	✓	✓
	Digital 2 (Coaxial)	✓	✓
	DAT In/Out (Coaxial)	✓	✓
Analog	CD	✓	✓
	Phono	✓	✓
	Line 1	✓	✓
	Line 2	✓	✓
	Line 3	✓	
	Tape 1	✓ (TAPE1/DAT2)	✓ (TAPE1/DAT1)
	Tape 2 (Monitor)	✓ (DAT1/TAPE2)	✓ (TAPE2/DAT2)
Function			
Cartridge Selector (MM/MC)		✓	✓
Bass Control		✓	✓
Balance		✓	✓
Headphone Output		✓	✓

Digital Pure-A II digital reference amplifier with K2 interface

Titanium-finished



- Power output: 2 x 100 watts*
- Digital Pure-A Type II amplification circuit
- K2 Interface
- Quadruple full-time linear 18-bit combination D/A converters
- Opt Super-A with Gm Driver

*at no more than 0.004% THD (8 ohms, 20 Hz — 20 kHz) (RMS)

AX-Z1010TN SUPER DIGIFINE COMPU LINK Component

Integrated Amplifier

Digital Pure A amplifier with 3 digital inputs

- Power output: 2 x 100 watts*
- Digital Pure-A Circuit for digital input signals
- D/A converter with 4-times oversampling digital filter built in
- Dynamic Super-A with Gm Driver for analog signals
- "D/A Converter Direct" function

*at no more than 0.003% THD (8 ohms, 20 Hz — 20 kHz) (RMS)

AX-Z911BK SUPER DIGIFINE COMPU LINK Component

Integrated Amplifier

Digital Pure-A circuit Type I (AX-Z911BK) /Type II (AX-Z1010TN)

"Digital Pure-A" circuitry realizes high-quality class-A operation as well as efficient high-power amplification when a digital signal is input, utilizing our advanced digital signal processing technology. Part of the input signal is used as a "prediction" signal which is used to analyze the level of the incoming signal while the main signal is delayed in memory before entering the D/A converter, in this way the gain in amplification is always optimum. The difference between the Type I and Type II systems is that in the Type I system, the amplifier is controlled by voltage, while in the Type II system, it is controlled by the bias current.

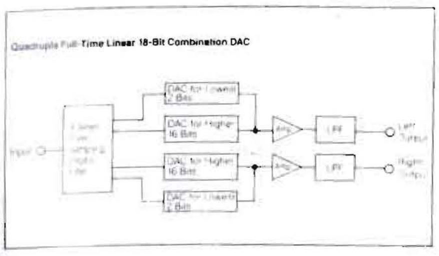
With the Digital Pure-A circuitry incorporated in the AX-Z1010TN and AX-Z911BK, the digital signal supplied directly from the CD player, etc. can be reproduced with extreme purity by giving it the ideal amplification. By utilizing the characteristics of digital signals, the incoming music signal can be stored in memory momentarily while the bias current or voltage required to optimize the amplification of the music signal can be calculated. As a result, the ideal high-power class-A operation is made possible with a relatively compact amplifier. In this way, the "pure class-A operation" necessary for extremely pure digital sound reproduction does not require a huge power transformer or an expensive, inefficient power supply with elaborate heat sinks. The resulting sound is pure, realizing the serious audiophile's dream of class-A operation.
The advantage of the Type II circuit over the Type I circuit is that as the time required for "prediction" is greatly reduced, the time lag which could be a serious drawback when the sound and picture in the playback of a video tape should be synchronized is negligible.

"K2 Interface" (AX-Z1010TN)

The "K2 Interface" — JVC's innovative digital signal transmission system — is also used in the AX-Z1010TN amplifier as well as our top CD player. The incoming digital signal (transmitted from the CD player, etc.) is supplied to the K2 Interface before D/A conversion so that only the required digital data (code components) is transmitted.

Refer to "Technical Notes" page 27 for more information.

By the use of an entirely new digital signal transmission system called the "code transmission system" employed in our "K2 Interface", in which the digital data is "recreated" according to the input data, "non-coded" components — resulting from ripple and jitter which may be introduced into the subsequent circuitry and greatly affect the resultant analog signal processing — are totally excluded. Since the K2 Interface completely shuts out the digital noise caused by such "non-code" components (ripple and jitter), any digital signals from the audio components connected to the AX-Z1010TN are applied to the D/A converter and subsequent analog sections, regardless of the quality of the digital component (so long as it is "digital"). Therefore, extremely pure digital "code" signals can be reproduced, exactly as the musician or producer intended, with ambience of the concert hall where the recording was made — which was never before possible.



Quadruple full-time 18-bit combination D/A converter with 8fs digital filter (AX-Z1010TN)

After the input digital signal is cleaned up by the K2 Interface, first the signal is over-sampled at a sampling frequency 8-times higher than the normal frequency (44.1 kHz in compact discs) in the digital filter, while the number of bits is also raised from 16 to 18. It is then converted to analog by the newly developed "combination" four DAC (digital-to-analog converters), in which the upper 16 bits and the lower 2 bits from the digital filter are processed separately and combined by a current adder; for higher fidelity, these operations are performed independently for the L and R channels.

The 8fs digital filter shifts the quantization noise to far higher frequencies so that it does not affect the audible frequencies, by multiplying the sampling rate up to 352.8 kHz, while the bit rate is raised to 18 bits. This 18-bit data is divided into two parts, and since the lower 2 bits which deal with lower level signals are processed separately, the D/A conversion accuracy is greatly improved while the linearity low to high signal levels is far better. This gives extremely accurate reproduction, especially of very low level signals.

Refer to "Technical Notes" page 32 for more information.

D/A Converter Direct function (All models)

The AX-Z1010TN and AX-Z911BK have built-in D/A converters which can accept digital signals directly from a CD player, etc. with a digital output. Even better, these amplifiers have a special facility called "D/A Converter Direct" which shortens the internal signal path to connect the signal from the built-in D/A converter directly to the output stage, as its name indicates. With this function ON, any circuits which could introduce distortion such as source selector switch or balance control are bypassed and the D/A converted signal is directly applied to the power amplifier via the master volume control alone.

With the D/A CONVERTER DIRECT switch on the front panel set to ON, the input digital signal from the digital output jack of the CD player, etc. is directly routed to the built-in D/A converter, then supplied to the power amplifier without passing through any circuitry which could affect the signals, except for the volume control. As a result, the total signal path is greatly reduced, maintaining pure signal transmission.

Three digital inputs including optical connector (All models)

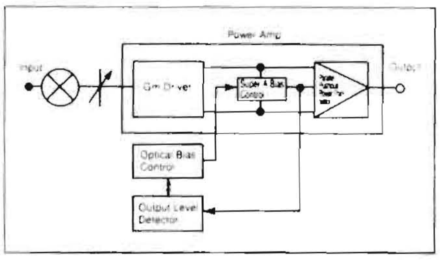
The AX-Z1010TN and AX-Z911BK are equipped with three digital input connectors; two coaxial and one optical. These allow the direct digital interfacing with any digital components such as CD players. Also, our digital reference amplifiers provide an additional "output" digital connector for recording with a future Digital Audio Tape deck, etc.

Since two types of digital connectors are provided, the AX-Z1010TN/AX-Z911BK can be connected to digital components with either a coaxial or optical digital output connector. The coaxial digital inputs permit digital interfacing with any digital components conforming to the Digital Audio Interface Format. If your CD player or other digital component has an optical digital output, digital signal transmission will be further improved by using an optical fiber cable. As there is no electrical transmission when optical interfacing is used, there will be no interference and therefore extremely pure sound.

Digital/analog separated construction (All models)

In the AX-Z1010TN/AX-Z911BK, the analog circuits are completely separated from the digital circuits, with one at the left and the other at the right of the amplifier. With the power supply section between them, this layout is designed to leave even more space at the center, between the digital and analog circuits. Furthermore, the mosl digital sections which could produce noise are completely shielded using solid plates.

With this circuit layout, any possible interference between the digital section and the analog section is greatly reduced. The result is purer transmission, of both the digital and analog signals.



"Opt Super-A" circuit (AX-Z1010TN)

In our efforts to achieve superior amplification, we haven't ignored analog input signals. The result of our consistent pursuit of better amplification is the "Opt Super-A" circuit — in which the bias current required for power amplifier to drive the speakers is controlled so that an optimum bias current is always supplied to the power amp stage, and this bias current is transmitted optically which gives the feature its name. The bias current is controlled optimally, according to the level of the input music signals so that an optimum value is always applied, to realize the class-A operation.

When the level of analog input signal varies, the "Opt Super-A" circuit immediately follows the change of input signal level to keep the amount of bias current supplied to the power amp stage sufficient for optimum operation. And since the output level detector always checks the output level, if the bias current is higher than it need be, it is reduced when the level drops. This control is made in the optical bias control circuit, in which optical isolation using a high-speed photocoupler prevents any feedback from the output stage. Thus more linear amplification with class-A operation is made possible with minimum switching distortion.

Refer to "Technical Notes" page 58 for more information.

Parallel push-pull power transistors (AX-Z1010TN)

In an amplifier, push-pull circuits are usually used to amplify the signals that drive the speakers. In this circuitry, the positive and negative portions of the waveform are amplified separately by two independent transistors before being combined to form the output signal. The output stage of the power amplifier section incorporates power transistors arranged in parallel, instead of the series connection used in conventional amps.

Since the power output transistors are arranged in parallel, more powerful amplification is made possible, and at the same time, the ability of the amplifiers to drive low-impedance speakers is greatly improved.

High-gain phono equalizer (All models)

A high-gain phone equalizer, using an active load in the first stage of the equalizer amplifier is provided to increase open-loop gain for use with either MM or MC cartridges.

You are not restricted to MM cartridges; you can also use a quality MC cartridge without connecting a separate transformer or head amp.

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Two-amp construction (AX-Z1010TN)

JVC's two-amp configuration is simplified structure with only two active stage block — the phono equalizer and the power amplifier. This two-amp design means that no separate tone control amp is used; the tone control circuit is part of the negative feedback network in the high-gain power amp.

This simpler design retains the integrity of the audio signals throughout the amplification process and ensures a better signal-to-noise ratio, for sound with greater clarity.

Connections for two tape decks (All models)

Two tape decks can be connected for dubbing in bi-direction and parallel recording.

With these terminals, dubbing is easy in both directions, TAPE 1 to TAPE 2 and vice versa; one source can be recorded onto two tape decks at the same time.

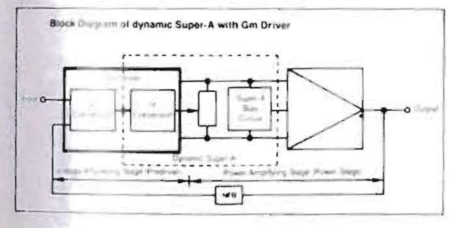
Dynamic Super-A

Dynamic Super-A circuit (AX-Z911BK)

Dynamic Super-A is a refinement of our revolutionary Super-A, the design that offers the silky, smooth, distortion-free sound of a class-A amp, and the efficiency of a class-B amp. Dynamic Super-A was developed with the singular purpose of catching up with the vast improvements that have been made in the dynamic range of new software technologies, such as digital mastering.

Low distortion, excellent open-loop response and minimized effects on the music due to counterelectromotive current fed back from the driven speaker are achieved and the dynamic noise and distortion that occur in actual in-use conditions are reduced

Block Diagram of dynamic Super-A with Gm Driver



Gm driver (All models)

Gm driver improves the real-life performance of an amplifier by driving the power stage at a constant voltage to reduce output impedance and achieve a flat frequency response.

The effects of the counterelectromotive force and variations in distortion due to the inherent non-linearity of power transistors are reduced.

JVC Amplifier Technology

In Pursuit of Class-A Amplification

Amplification techniques and why class-A operation is superior

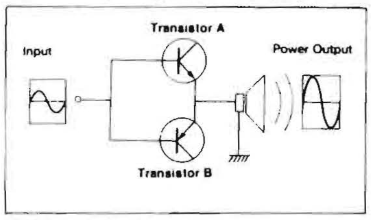
Before introducing the Digital Pure-A II and Opt Super-A circuit, some basic knowledge of amplifier design is necessary, including class-A and class-B amplification. For efficiency, Class-B amplifiers use push-pull circuitry with independent transistors amplifying the positive and negative halves of the signal, then the outputs of the transistors are combined to drive the speakers. In this way, when the signal is positive, the transistor that amplifies negative parts of the signal is off and vice versa. The problem occurs at the point where the two signals are combined, where switching distortion occurs.

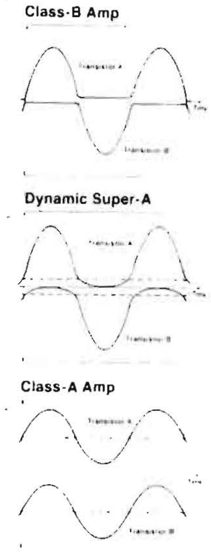
In "non-switching" amps including our Dynamic Super-A amplifiers, to eliminate switching distortion, a small amount of bias current is applied to both transistors at all times, so that they are never switched off. In this way, when the waveforms are combined, the result is much smoother.

Class-A amplifiers, on the other hand, are far less efficient as both transistors are driven at full power at all times. The output from each transistor is symmetrical with the input signal and this is also true of the waveform after the outputs are combined. For this reason, class-A amplifiers reproduce sound with the lowest possible distortion which makes them ideal for the reproduction of music and, if it wasn't for their low efficiency, they would be the choice of any true audiophile.

JVC developed "Opt Super-A" for analog sources and "Digital Pure-A Type II" for digital sources to further improve amplifier design.

Output Signal of Power Amp





Operating current waveform of transistors A and B

I. Opt Super-A circuit

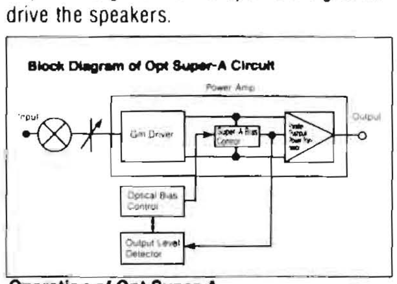
"Opt Super-A" — another milestone in amplification of analog sources

As shown in the diagram, in the Opt Super-A amplification circuit, the bias current supplied to the power transistors is controlled so that they have the optimum operating condition for the music signals supplied at any instant Part of the constant-voltage signal from the Gm driver is supplied to the Super-A Circuit and from there to the Signal Level Detector. This produces a DC signal which is proportional to the level of the music signal; this is fed to the Optical Bias Controller which, in turn, supplies a bias-control signal to the Super-A Circuit, to optimize the operation of the power stage.

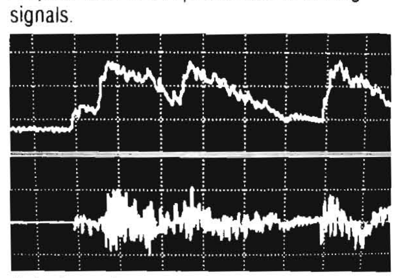
The Optical Bias Controller uses a photocoupler for electric isolation, so that there is no feedback loop which could degrade the performance of the amplifier.

As shown in the diagram, the input music signals are monitored to detect the peak level of the signal, and this detection signal is used to vary the bias current, so that an optimum bias current is applied to generate the signals required to drive the speakers at all input signal levels. With this operation, as shown in the figure, the bias current follows the input signal, so that the output stage is supplied with the bias required to generate the optimum signal to drive the speakers.

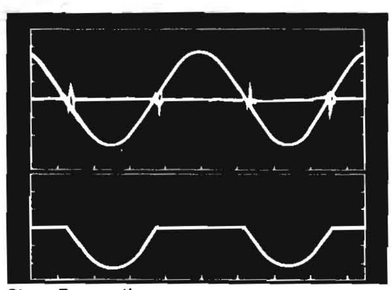
Even in the age of digital sound, JVC is doing everything possible to improve the amplification and reproduction of analog signals.



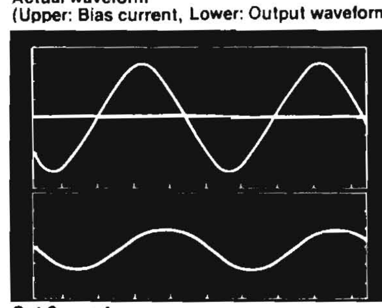
Operation of Opt Super-A



Actual waveform
(Upper: Bias current, Lower: Output waveform)



Class-B operation



Opt Super-A

II. Digital Pure-A Type II

Outline of "Digital Pure-A" Technology

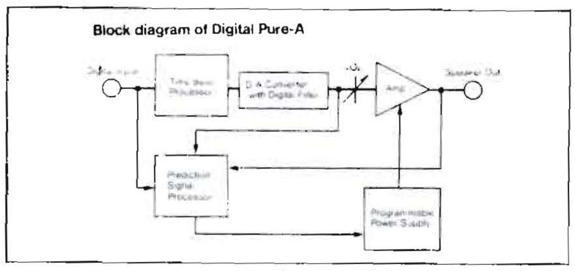
JVC engineers used the fact that with digital program sources the signal is processed digitally to make possible class-A operation with a lower power consumption and the generation of less heat, for higher efficiency.

By including the D/A converter in the amplifier, digital signals can be input directly to the amplifier. With the signal in digital form, it is possible to control the delay time without any degradation of the signal. This characteristic of digital signals was employed in the "Digital Pure-A" amplifier, so that it could provide high-power Class-A operation at all time, but without the low efficiency inherent in Class-A amplifiers.

Digital Pure-A amplifiers use "signal prediction", possible only with input digital signals, to maintain Class-A operation. The input signal is divided into two parts; one is stored temporarily in a large capacity memory called a DRAM (dynamic random access memory) in the Time Base Processor, and the other part is used for "prediction".

In signal prediction, the level of the signal read into memory is measured to judge just how much amplification is required. In this way, the transistors in the amplification stage can be supplied with exactly the right voltage needed to amplify the signal read out from memory.

Block diagram of Digital Pure-A



How is prediction used to increase power output?

Even when the music seems to be at a relatively high volume, amplifiers are only required to deliver their highest power for short periods. The AX-Z911BK which incorporated a Digital Pure-A circuit normally operated as a class-A amplifier with an output of 20 watts per channel. When a higher level of amplification was required, the voltage supplied to the power amplifier stage was increased for class-A amplification with an output of 100 watts per channel. In this way, the optimum voltage was supplied with high as well as low input signals.

In the AX-Z911BK, the music signal was delayed for 150 msec in the Time Base Corrector while switching of the power supply voltage was done 120 msec before the signal was read out of memory, allowing a sufficient margin for the switching of the power supply.

2nd generation — "Digital Pure-A II"

"Digital Pure-A II" also uses signal prediction, but in a different way; the digital signal prior to D/A conversion is used to control the bias current, rather than the voltage, for more effective operation of the power amplification stage.

As shown in the block diagram, the encoded digital signal from the source component, after being decoded, is divided into two parts; one is supplied to the Time Base Processor in which the incoming digital signal is stored for approx. 10 msec before entering the "K2 Interface" and subsequent D/A Converter, etc., and the other to the Prediction Signal Processor to be used for prediction, to control the bias current supplied to the power amplifier which drives the speakers.

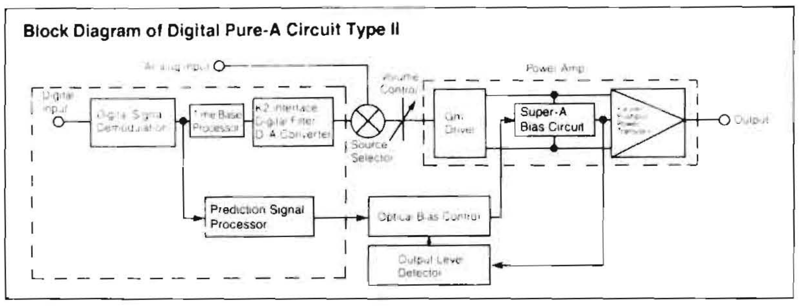
The Prediction Signal Processor generates a signal which is proportional to the instantaneous peak level of the music signal; this is supplied to the Optical Bias Control Circuit which supplies the bias control current to the Super-A Bias Control Circuit, to optimize the operating condition of the amplifier. The Optical Bias Control Circuit incorporates a photocoupler for electrical isolation, so that there is no feedback loop that could degrade the operation of the amplifier. When the higher bias current is not required, the Super-A Bias Control Circuit feeds a signal to the Signal Level Detector and Optical Bias Control Circuit which gently attenuates the bias.

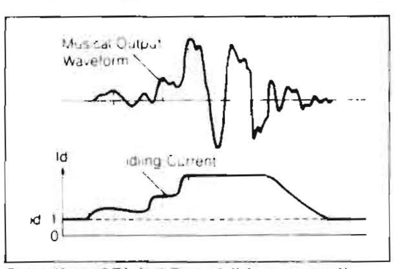
As shown in the diagram, the bias current is raised in two stages, just before the rise in the music signal. This shows that "prediction" is performed approx. 10 msec. immediately before the high-level music signal arrives, with sufficient bias current always supplied according to the level of the music signal. The resultant output current will be smooth as shown in figure below, as required for class-A operation.

Digital Pure-A, because the prediction signal controlled voltage, required the charging of a large capacitor and a relatively high current, and this resulted in a much longer time delay (approx. 150 msec).

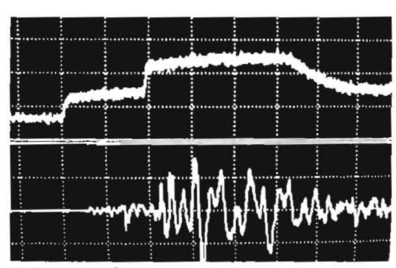
By controlling the bias current instead of the drive voltage, Digital Pure-A II reduces the time delay to 10 msec — an improvement over Digital Pure-A.

Block Diagram of Digital Pure-A Circuit Type II





Operation of Digital Pure-A II (conceptual)

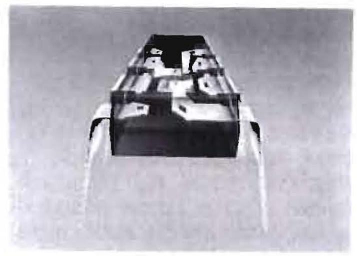


Actual waveform
(Upper: Bias current, Lower: Output waveform)

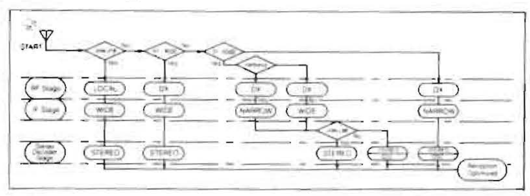
AMPLIFIERS

Feature Highlights of '90 Tuners

1 ***Our "Opticalink" optical transfer system completely eliminates digital/analog interference***



2 ***The computer-controlled "reception servo" system maintains optimum reception conditions at all times***



Feature Comparison Chart

	FX-1010TN	FX-1100BK
Circuit		
Opticalink	✓	✓
Digital Synthesizer (FM/AM)	✓	✓
Reception Servo	✓	✓
2-Antenna Inputs	✓	✓
Function		
Preset Stations (Random)	40	40
Station Name Memory (6-column)	✓	✓
Variable Stop Level	✓	✓
Program/Auto/Manual Memory	✓	✓
Preset Scan	✓	✓
Preset Cancel	✓	✓
FM IF Selection	✓ (Narrow/Wide)	✓ (Narrow/Wide)
FM RF Selection	✓ (DX/Local)	✓ (DX/Local)
Auto QSC	✓	✓
Rec Calibration	✓	✓
Display		
dB Indication	✓	✓
Display Panel	FL	FL
Alphanumeric Display (6-column)	✓	✓
COMPU LINK Component	✓	✓

Computer-controlled digital synthesizer tuner with "reception servo" system

Titanium-finished

FX-1010TN SUPER DIGIFINE COMPU LINK Component

Computer-Controlled Digital Synthesizer Tuner

- Computer-controlled "reception servo" system
- "Opticalink" system
- 6-character station name display
- Direct-access numeric keypad
- Random preset memory for 40 FM/AM stations

Computer-controlled digital synthesizer tuner with "reception servo" system

FX-1100BK DIGIFINE COMPU LINK Component

Computer-Controlled Digital Synthesizer Tuner

- Computer-controlled "reception servo" system
- "Opticalink" system
- 6-character station name display
- Direct-access numeric keypad
- Random preset memory for 40 FM/AM stations

Feature reference

Technology/Function	*Benefits*

"Reception Servo" (All models)

Technology/Function	Benefits
Computer controlled reception servo system automatically selects the operating mode of the front-end, IF and multiplex decoder stages to provide optimum reception considering the degree of interference and the strength of the tuned signal.	Tuning is always optimized so the best possible sound is obtained regardless of your location.

Computer-controlled operation (All models)

Technology/Function	Benefits
Operations are extremely convenient thanks to the computer; auto memory, preset memory and preset scan are available. In addition, auto QSC, dB-referenced signal strength indicator and variable stop level functions are available.	Tuning operations are easy and accurate, so even if you do not know the frequency of a station, you can tune to it exactly.

"Opticalink" (All models)

Technology/Function	Benefits
This system electrically decouples the digital section (microcomputer and display) from the analog circuitry and transmits all signals using a photocoupler.	A variety of noise from digital circuits transmitted through the power supply line and ground line is shut out and only pure signals are transmitted as they are converted to light.

IF bandwidth select function (All models)

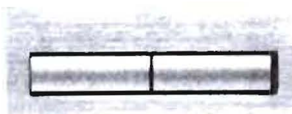
The IF bandwidth select function using the computer lets you tune to broadcasts in a more listenable condition.

When a station you are attempting to listen to is subject to interference from another, stronger station, the computer automatically selects the narrow IF bandwidth to sharpen selectivity and reject interference. Your desired station will now be received loud and clear. On the other hand, if there's no interference, the IF band-width remains wide, so you can enjoy the best possible hi-fi sound.

Auto memory functions (All models)

Auto memory functions using the computer to automatically preset stations one after another, with the user selecting the starting frequency, even in the middle of the tuning band.

This function is extremely convenient presetting stations in optimum condition wherever you live, eliminating your having to check broadcast frequencies.



Program memory and program monitor (All models)

Program memory and program monitor let you program several broadcasts and check the programming for unattended recording.

When used with a multi-programmable timer, the program memory offers true convenience: it automatically tunes to different stations held in the memory each time the tuner is turned on and off, as controlled by the timer.

Preset scan function (All models)

Preset scan function lets the user automatically tune to each of the preset FM (or AM) stations in memory and hear them for five seconds each.

This is a very convenient way to search for a station you want to hear, letting you check the programming offered by your preset stations at any time.

Preset memory of 40 FM/AM stations (All models)

Preset memory function lets you preset 40 FM and AM stations.

Once a station is preset, you can instantly recall it by simply touching a button; a total of 40 FM and AM stations can be preset in any order.

Preset Cancel function (All models)

This function is used to delete preset channels in which no broadcast station frequency is stored, from the preset memory. With this, unused preset channels will be skipped during preset scanning operation.

Although our tuners provide up to 40 station presets for FM and AM broadcasting, sometimes all of these are not always required in actual use. In such a case, unused (or temporarily unwanted) preset channels can easily be "cancelled" from the internal memory, while only the required preset channels are held in memory. This function could be convenient in an area where only a limited number of broadcasts are available.

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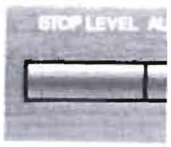


Variable stop level function (All models)

This lets the user determine the muting threshold level in steps of 5 dB over a 20 to 60 dB range for FM and a 60 to 90 dB range for AM; this setting is effective only for stations held in auto memory.

Select a higher level and you'll only get powerful, good-sounding stations while muting out stations with poor sound. By choosing a lower level setting, all stations, unless they are extremely weak, will be received. This feature is very handy when there are many stations to choose from.

dB-reference signal strength indicator (All models)

dB-reference signal strength indicator displays the strength of tuned stations in dB.

This feature is useful when orienting the antenna for optimum reception, etc. It also lets you check whether a station can be received satisfactorily or not in your area.

Auto QSC (All models)

The auto quieting slope control (QSC) circuit operates effectively when a stereo broadcast with a weak signal strength is received; when the signals is lower than 39 dB, this circuit is switched to on automatically and reduces noise by controlling the L and R components of the sub signal, resulting in a 6-dB improvement in the stereo signal-to-noise ratio.

Stereo broadcasts can be received with more listenable sound because stereo noise is removed when the signal from a stations is weak.

Station name display (All models)

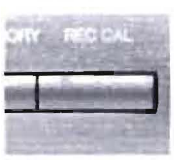
A 6-figure display (letter and numerals)) is used in the FX-1010TN/FX-1100BK, making it possible to assign up to six alphanumeric characters for each preset station.

The user can input letters as required, for example, "JAZZ-3" for a third jazz-oriented station.

2 antenna inputs (All models)

2 antenna inputs are provided for the connection of two antennas which can be selected.

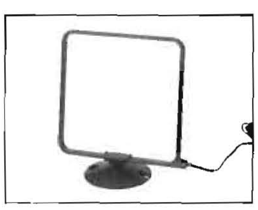
You can point these two antennas towards different stations for the best possible reception of different broadcasts. In addition, each preset position can be programmed to be received from either antenna "A" and "B", so that as a station is selected, it is automatically received from the optimum antenna.



Record calibration signal generator (All models)

Record calibration signal generator outputs a standard 400 Hz signal for recording level adjustment.

You can easily set the recording level for different broadcasts or types of tape.



Separate AM loop antenna (All models)

A separate AM loop antenna with a stand is provided and can be placed in the location that gives the best reception.

The user can move the AM loop antenna to find the position for optimum reception. The antenna can be installed anywhere; this flexible antenna setting is as convenient as a portable radio that can be placed anywhere.

COMPU LINK

JVC COMPU LINK control system

With JVC's COMPU LINK control system, various COMPU LINK components interact with each other via a common "bus"

It has two extremely important functions; one is "automatic source selection" allowing simplified operation when the input source is switched, and the other is "synchro recording" which uses a convenient synchronized system to start and stop recording/editing.

And if one of the COMPU LINK components incorporates the COMPU LINK remote control feature, they can all be operated with the single remote control unit supplied with the COMPU LINK remote control component.

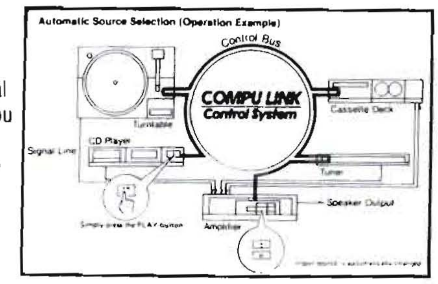
For greater flexibility, with the unified A/V remote control unit provided with JVC receivers, certain JVC video components — VCRs and TVs — can also be controlled directly.

So, if you build a hi-fi system around JVC COMPU LINK components, you can easily upgrade it to a comprehensive A/V system with a unified remote control by simply adding a COMPU LINK remote control component.

■ *Automatic Source Selection*

One of the most convenient features of the COMPU LINK system is its ability to switch inputs and play the required source component, by simply touching a button on either the amplifier/receiver, the source component itself, or the remote control.

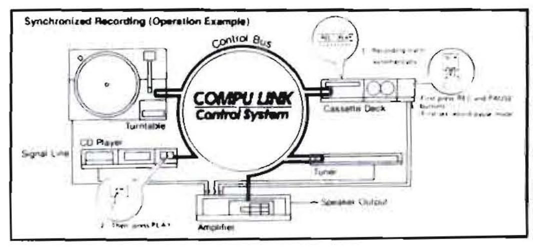
For example, if you want to listen to a compact disc while you're playing a cassette tape, all you have to do is touch a single button — the CD button on the amplifier, or the PLAY button on the CD player. Whichever you press, the tape stops, the CD player starts, and the signal path is automatically switched over so you hear the CD. This automatic sequence works in the same way with other source components.



■ *Synchronized recording*

Another convenient feature of the COMPU LINK control system is that it synchronizes the operation of the cassette deck with the CD player or turntable, so you can dub from compact discs or records to cassette tapes, by simply pressing a single button.

To dub from a compact disc to cassette tape, for example, first set the deck to the record-pause mode, then select the tunes on the compact disc that you want to record. Now, press the PLAY button of the CD player: The required tunes will be automatically recorded on the cassette deck. And when the CD player finishes playing, the cassette deck also stops automatically. By programming the compact disc to play tunes in a certain order, it's easy to record a customized tape; when you use this feature, the gaps interval between tunes on the tape required for Music Scan and other playback functions to operate correctly will be left automatically.

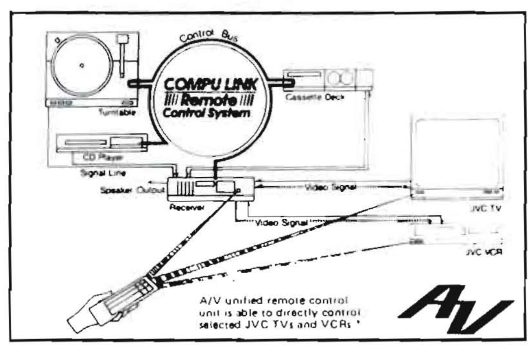


■ *COMPU LINK remote control system*

The COMPU LINK remote control system goes one further than the conventional COMPU LINK control system. With it, most JVC components respond to signals from a COMPU LINK remote control unit. All the operations described above are possible with the remote control, putting complete control of your hi-fi system in the palm of your hand.

Even if COMPU LINK components are not designed for remote control on their own, when they are connected to COMPU LINK Remote Control Components with remote cables, remote controlled operation is possible. At this time, the remote control signal is transmitted to the required component via the remote cable providing the signal bus.

This gives you complete control of your CD player, cassette deck, tuner, DAT deck, SEA graphic equalizer, and other components, from your listening position.



■ *Integrated Audio/Video remote control*

Some COMPU LINK remote control components are supplied with a unified A/V remote control unit which works with both audio COMPU LINK components and certain JVC video components (such as VCRs and TVs). Audio components are controlled via the COMPU LINK remote component, while the video components are controlled by signals directly from the remote control; these signals are received by remote sensors on the front panels of the video components and allow you to call up any TV channel directly, tune to TV channels in sequence, and operate the VCR for recording and playback, etc.

Further enhancements of the COMPU LINK control system

■ CCS: COMPU LINK Communication System

For greater convenience, we've introduced an interactive display feature called the "COMPU LINK Communication System", or CCS, which indicates the source selected and operating mode using an information-packed alphanumeric and pictorial display on the front panel of certain receivers.

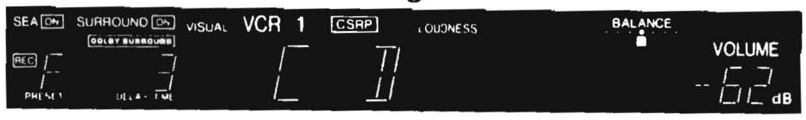
For example, with JVC's new line-up of receivers, when an FM broadcast is selected with the remote control, the receiver's display first shows the selected source "FM", then it displays the station name you registered for that station. In this way, new JVC COMPU LINK components make possible interactive communications.

These interactive operation make JVC's COMPU LINK system amazingly convenient, and give it a vast potential for future expansion.

■ CSRP: COMPU LINK Source Related Presetting

This year's top-end receivers are provided with "CSRP", COMPU LINK Source Related Presetting, which allows the user to save all the settings (volume, balance, etc.) and parameters (for S.E.A., the surround processor) for each source independently, to be recalled whenever that source is selected.

With CSRP, users no longer have to readjust or modify settings when they select a particular source.



Whenever the source is selected, the various settings will be shown on the display, in sequence; finally the display shows the source and other important settings. These settings can then be modified using the remote control provided with the receiver, to achieve the required sound field.

With CSRP, our new receivers are no longer just a amplifier that incorporates a tuner, but the control center of the user's audio/video component system.

COMPU LINK Component

This logo indicated a component that incorporates JVC's COMPU LINK control system. When used with other COMPU LINK components, this allows interactive operations such as one-touch automatic source selection and playback, and coordinated operations such as synchronized recording.

CD players

XL-Z1010TN/XL-Z611BK/XL-Z431BK/XL-Z331BK/XL-V231BK/XL-V131BK/XL-V85BK/XL-G512NBK XL-M701BK/XL-M403BK/XL-M303BK/XL-M87BK/XL-R202BK/XL-R86BK

Amplifiers

AX-Z1010TN/AX-Z911BK

Tuners

FX-1010TN/FX-1100BK/FX-87BK

Cassette decks

TD-V1010TN/TD-V711BK/TD-V531BK/TD-R431BK/TD-X331BK TD-W901BK/TD-W803BK/TD-W503BK/TD-W303BK/TD-W203BK/TD-W87BK/TD-W85BK

Turntables

AL-FQ555BK/AL-F353BK/AL-F97BK

COMPU LINK //// Remote //// Control Component

This logo indicates a component which can work as the center component of a remote control system, enabling total remote control of the components connected to the unit. COMPU LINK remote control components include remote controlled amplifiers, receivers and cassette receivers.

Amplifier

AX-R87BK

Receiver

RX-1010VTN/RX-903VBK/RX-803VBK/RX-703VBK/RX-503BK/RX-403BK/RX-R85BK

COMPU LINK/ SYNCHRO terminal

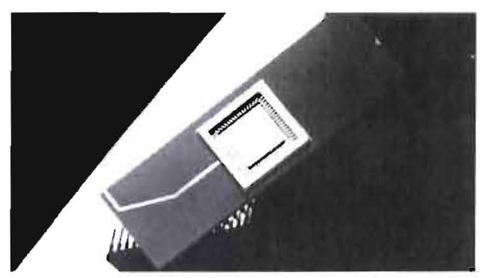
COMPU LINK components are equipped with SYNCHRO terminals which should be connected to the SYNCHRO terminals of other COMPU LINK components using exclusive synchro cables (bus cables) provided; these are the control buses between the components that make interactive operation possible.

DIGITAL ACOUSTICS PROCESSORS

Feature Highlights of '90 Digital Acoustics Processors

1 ***JVC's advanced symmetrical 6-point sound field analysis with specially designed measurement microphones used to determine parameters***

2 ***There are 20 programmed and 20 user-programmable sound field patterns***



3 ***Independent setting of five acoustic parameters and four source/room control parameters***

6-channel full-digital ambience control digital acoustics processor with remote control

Titanium-finished

XP-A1010TN SUPERDIGIFINE

Digital Acoustics Processor

- 16-bit quantization with sampling rate of 48 kHz
- Independent setting of five acoustic parameters and four source/listening room parameters
- 20 preset and 20 user-programmable sound field patterns
- 6-channel ambience operation with remote control
- Symmetrical 6-point sound field analysis

6-channel full-digital ambience control digital acoustics processor with remote control

XP-A1000TN SUPERDIGIFINE

Digital Acoustics Processor

- 16-bit quantization with sampling rate of 48 kHz
- Independent setting of five acoustic parameters and four source/listening room parameters
- 20 preset and 20 user-programmable sound field patterns
- 6-channel ambience operation with remote control
- Symmetrical 6-point sound field analysis

Digital Acoustics Processor Description Story

1 *Propagation of Sound in Concert Hall*

When we listen to the music in the concert hall, the sound we hear is not only the sound directly from the instruments or the singers, but also there are many reflections from the side walls, floor and the ceiling. (See Fig. 1.)

The sound is made up from three major components; direct sound (DS), early reflections (ER) and reverberations (REV), as shown in Fig. 2.

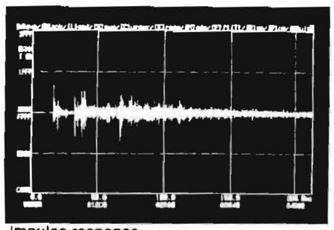
Among these signal components, the key to the unique acoustic characteristics of any auditorium are the early reflections, generated by the direct, "original" sound, reflected one or more times from the walls and ceiling. Each of the reflections has a different level and appears to come from a different direction (in other words, they have individual "vectors"), while the pattern of early reflections is different for different concert halls, auditoriums and jazz clubs, wherever music is performed.

At the same time, each acoustic space (concert hall, etc.) and home listening room has a unique pattern of reflections and reverberations, mainly caused by these "early reflections", which differ depending on the size of the hall (or listening room), its interior furnishings, etc. Some rooms are "live", with many reflections, and some are acoustically "dead".

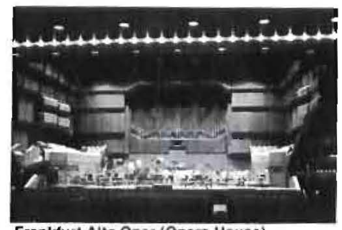
As shown in Fig. 3, the sound generated on the stage is propagated in all directions. Some of the signals reach the listener's ears directly, and some are reflected from the side and rear walls and ceiling before reaching the listener.

As above, from our study and analysis, we found that the pattern of the early reflections greatly depend on the type of acoustic space (concert hall or listening room), mainly the shape of the space and the characteristics of the wall materials.

For example, as shown in Fig. 4, when the sound is generated in a wide space with walls with less absorbing characteristics, there is not so much difference in level between the direct sound and the early reflections, while the interval between reflections is longer. When the sound is generated in a narrow space with walls that absorb sound, the gradient with which the level decays steep and the interval between the reflections becomes shorter.



Impulse response



Frankfurt Alte Oper (Opera House)

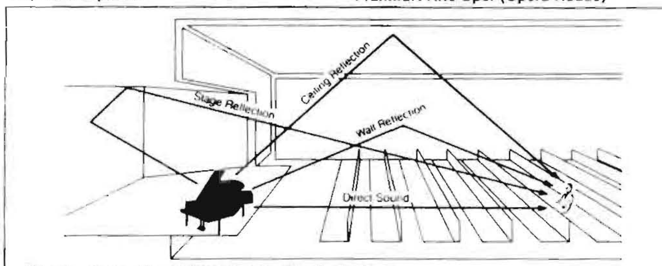


Fig. 1 Acoustic response of a musical place

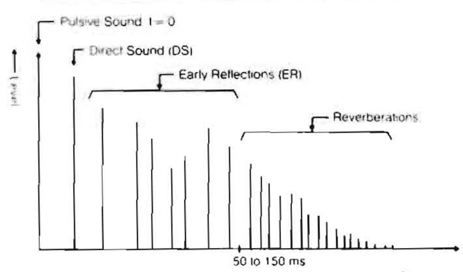


Fig. 2 Typical pattern of sound

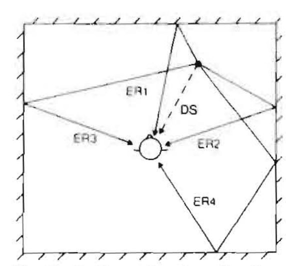


Fig. 3 Pattern of early reflections

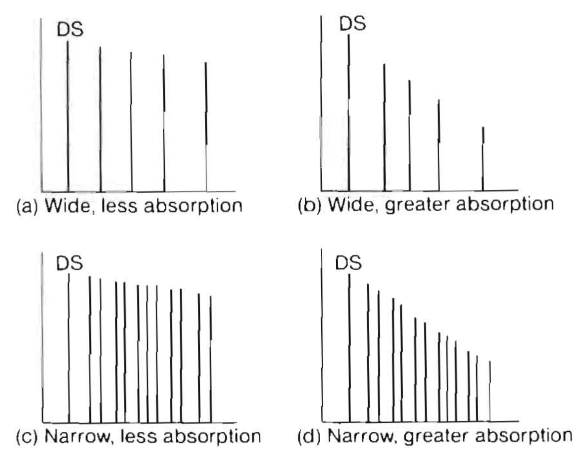


Fig. 4 Early reflection patterns and acoustic characteristics of different types of room

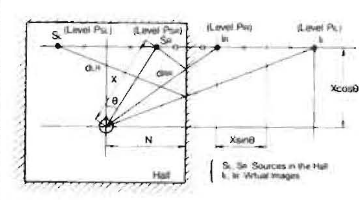
2 Virtual Image Sources

In Fig. 4, when a certain reflection is heard by the listener, the sound source seems to be at the position far beyond the walls, as shown in Fig. 5.

As shown in Fig. 5, one component of the sound generated from the sound source is reflected from the back wall of the stage, and then reflected from the side wall before it reaches the listener's ears. In this case, the listener perceives the sound source as coming from point I, far outside the concert hall (listening room). This means that, from what we hear, the sound source is at that position, in the sense of level and the direction from which the sound comes. — This is called the "virtual sound image".

We found that distribution of the virtual sound images is different for different concert halls, etc. Therefore, if the distribution of virtual sound images in famous concert halls, etc. are measured and analyzed, their acoustic characteristics could be analyzed.

Fig. 5



Relationship between actual sources and virtual sources

3 Newly-developed Symmetrical 6-point Sound Field Analysis Method

To measure the various acoustic parameters of places where music is played — concert halls, churches, jazz clubs, etc., — done in conjunction with the development of our Digital Acoustics Processing System, we developed a new high-precision measurement system called the "symmetrical 6-point sound field analysis method". This innovative computer-aided measurement system uses a specially designed measuring instrument consisting of three pairs of microphones installed parallel to each other; it measures the acoustic properties of different listening environments accurately and precisely.

In this system, microphones are installed symmetrically at one position, in the directions of the X, Y and Z axes. The sound sources used for measurement are very short pulsive sounds using three pistols used at the center, left and right of the stage. This use of a number of sources reveals the greater differences in the propagation of the sound and the reflections, depending on the position of the source.

Symmetrical 6-point Sound Field Measurement microphone

4 Distribution of Virtual Image Sources

We measured the sound field characteristics of concert halls and other places where live music is performed using this new system. In this way, actually measured data was used for the total analysis of the ambience parameters, using an advanced computer system.

Data from various measurements were input to an advanced computer-aided analysis system, and these results are summarized in "Virtual Image Source" diagrams.

One shows "impulse response" to the pulsive sound, another, the distribution diagram of "virtual" sources, and the third, the three-dimensional distribution of "virtual" sources.

Fig. 6-a shows the measured results of impulse response when a pulsive sound (like a pistol shot) is generated on the stage, showing the pattern of decay in a specific concert hall. And Fig. 6-b shows the two-dimensional distribution pattern of sound images corresponding to a pulsive sound source, obtained in the same concert hall, while Fig. 6-c shows the three-dimensional pattern of the same results.

In these diagrams, the point where the axes cross at the center of the figure is the listening position and the box is the hall; the largest circle shows the direct sound and the smaller circles show "virtual" sound images, from which the listener perceives that the sound originated. The sizes of the circles show the relative levels of the signals and the centers of the circles show the positions of the source of direct sound and the apparent sources of early reflections.

From the results of our extensive study using data measured in various locations, we found that the distribution of virtual images changes greatly depending on the size of the auditorium and on the points at which the measurement was performed. And at the same time, we also found that the distribution of virtual images changes when the point source shifts to the left or right of the stage.

As can be seen from Fig. 6, in addition to the direct sound (DS) which reaches the listener directly, he/she also hears reflections at different levels, from different directions. The distribution of these early reflections, that is the virtual sound sources, is shown in the diagrams which were compiled from the measurement of sound in specific halls.

The diagrams in Fig. 6 show the distribution of virtual sound images in a large concert hall where classical concerts are generally performed, while Fig. 5 shows the sound images in a typical small jazz club with a low ceiling,

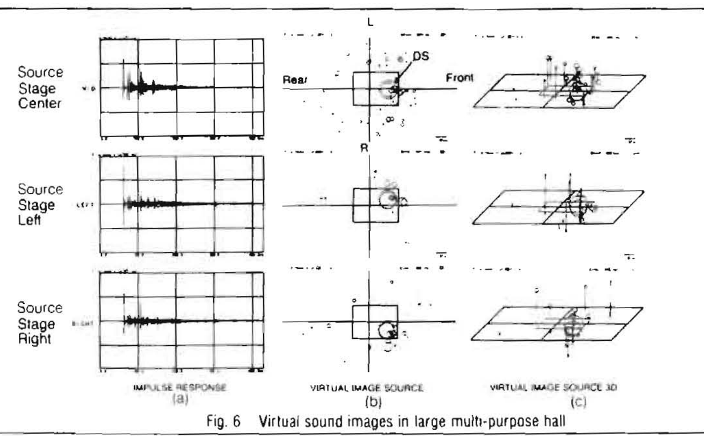


Fig. 6 Virtual sound images in large multi-purpose hall

5 *Digital Acoustics Processing for the Ultimate Sound Field Reproduction*

The acoustic characteristics of various sound fields — in concert halls, etc. — were measured using our exclusive "symmetrical 6-point sound field measurement microphone". All the measurements were performed with these special microphones placed at the best seat in a number of famous concert halls and places where music is performed, around the world.

To simulate the sound field in any of these places, the directions and levels of reflections and reverberations of the hall, etc. must be recreated accurately.

But if the acoustic parameters where the recording was made and the listening room where the music is reproduced — each has its own characteristics — are not taken into consideration when reproducing the ambience of a hall in a home listening room, excessive reflections and reverberations may totally eliminate any sense of realism.

This is because, if the source sound (Compact Disc, etc.) already includes the echoes and reverberations of the concert hall, or if these effects have been added intentionally by the mixing engineer, etc. to give acoustic presence, the programmed data will add further reverberation, and this could result in an over-processed sound field that is exaggerated.

Or, if the user's listening room has solid walls, the preset parameters could add too much reverberation to the original sound.

To prevent these problems, the conditions when the source was recorded and those of the listening room were also taken into consideration so that a more accurate sound field could be reproduced.

As a result, we've developed our exclusive digital acoustic processing system, by considering all the possible factors that might be related to recreating sound fields with extreme accuracy. The culmination of this research is the XP-A1010TN Digital Acoustics Processor — the crystallization of our long experience in all fields of audio and results of our determination to recreate "real" sound fields, by using a number of advanced digital technologies.

It has been achieved by the development of the three Digital Acoustics Processing VLSIs, custom designed by JVC, and ROM chips which contain data derived from actual measurements, in addition to highly accurate A/D and D/A conversion circuitry.

The input source signal is A/D converted into a digital signal, using 16-bit quantization with a sampling rate of 48 kHz, with a 64-times oversampling twin A/D converter, while the D/A conversion of the processed data is done by a 16-bit D/A converter with a 4-times oversampling digital filter.

With this chip set, all signal processing is done in stereo, and the 2-channel input signals are processed as they are.

Since all the signal processing is done digitally, signal deterioration and degradation are minimized, while all these operations are performed under microcomputer control for assured accuracy. With the XP-A1010TN, the user can easily recognize that the sound field is expanded, for a listening experience that's more exciting than ever.

Custom-designed Digital Acoustics Processing LSI

6 *Actual Operation with the XP-A1010TN*

With the JVC XP-A1010TN, the user can experience the "true" presence of certain famous concert halls, clubs, and even a cathedral, etc. in the user's own listening room, by recreating the sound field of the specified location.

The XP-A1010TN comes with the 20 sets of parameters programmed for recreating famous concert halls, etc.; in addition, it allows the user to store up to 20 sets of user-programmed patterns, with names the user can preset.

At the same time, the "listening room condition" and the "sound field condition" can be adjusted in detail, enabling extremely accurate reproduction of sound fields, set exclusively for the user's own listening room.

For the most accurate possible sound field recreation, the XP-A1010TN requires two pairs of "sound field" speakers with the amplifiers driving them, in addition to an existing 2-channel stereo hi-fi system.

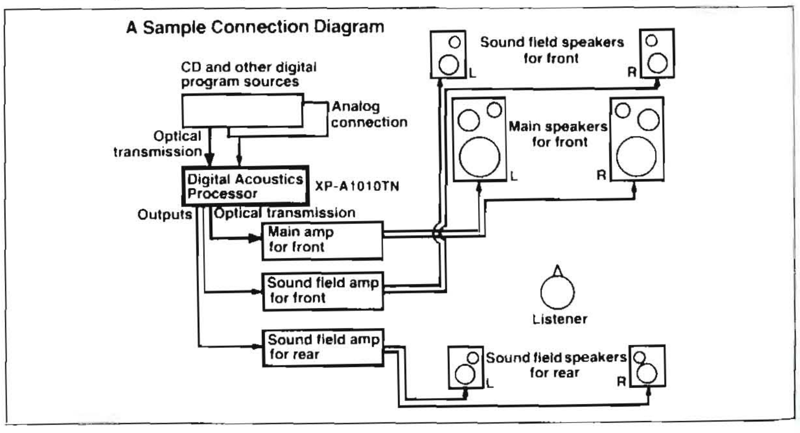
The most effective speaker placement, which gives the optimum effect, is with the four additional speakers at the four top corners of the room, with two at the top of the wall facing the main listening position, and two behind. This placement, with the speakers above the listener is best, and it is possible because the speakers required to give ambience can be relatively compact.

However, for convenience and economy, a simplified system is possible, using a single pair of additional speakers. Although the XP-A1010TN is designed as a 6-channel processing system, it can also be used in a 4-channel configuration. In this case, the sound which is to be reproduced by the front "sound field" speakers is combined with the front channel signals and output from the front-main speakers.

All the signal processing is done digitally in the XP-A1010TN. So that digital signals can be input with less distortion, the XP-A1010TN has two optical/coaxial digital inputs for the direction connection of CD players, etc. as well as inputs and outputs for the connection of a DAT deck.

All parameter settings and operations can be controlled from the listening position, using the remote control provided with the XP-A1010TN, while parameter settings, etc. are indicated on the FL display on the front panel, for operation convenience and flexibility.

A Sample Connection Diagram



7 Superiority of this JVC-exclusive Method

1) We used a more accurate measuring method:

All the measurements of the various concert halls, etc. were performed using our innovative "symmetrical 6-point sound field measurement microphone" rather than the "closed 4-point" system used in another manufacturer. At this time, we use pulsive sources placed at three positions on the stage; stage left, stage right and at the center, whereas the other system used a single source.

This makes possible stereo signal processing instead of the other system's monaural signal processing, for more accurate sound field measurement.

2) Sound field analysis includes listening room and source condition:

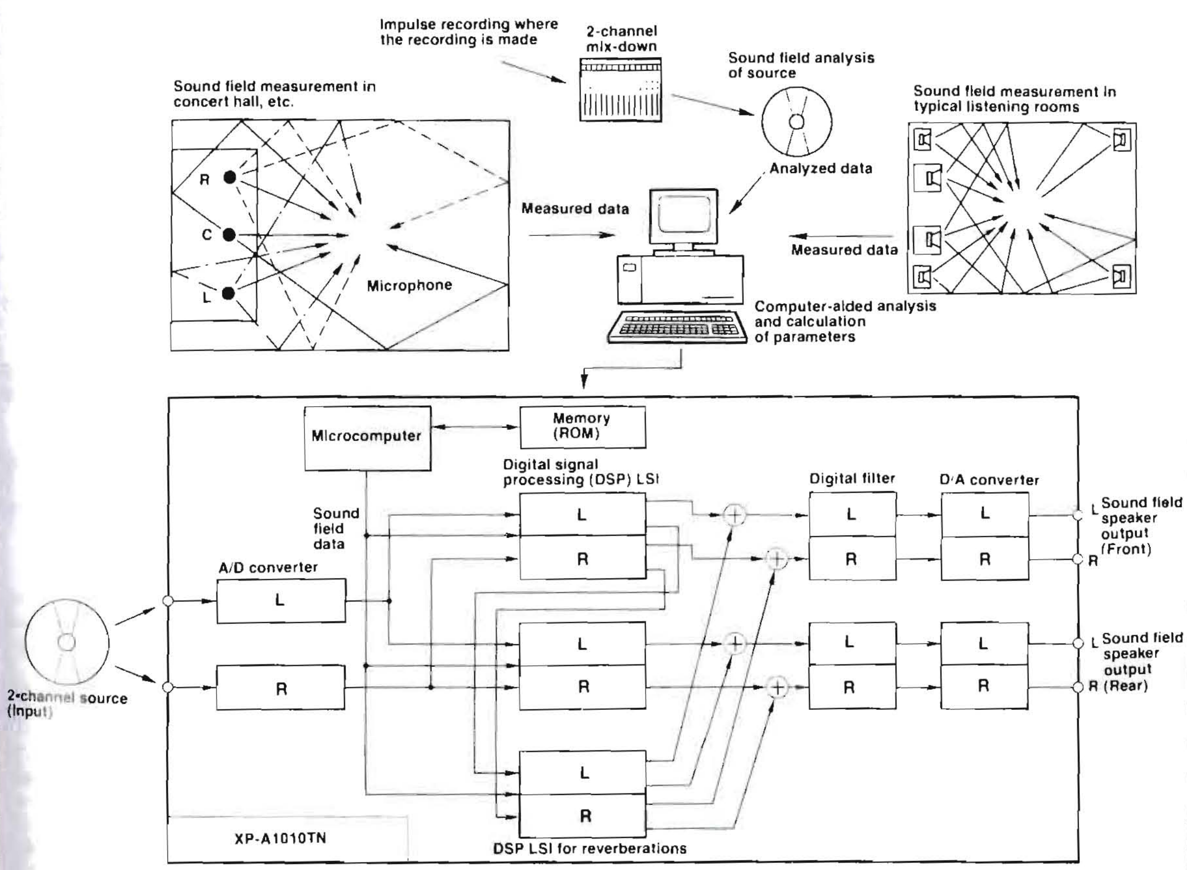
In addition to the acoustic characteristics of the concert hall, etc., we also include data derived from the analysis of listening room and source conditions, such as recording studio or hall, etc., while the possible acoustic characteristics of the listening room are also considered
To accomplish this, the XP-A1010TN)TN has adjustable parameters for listening room compensation as well as for source conditions, making more accurate and "natural" sound fields possible.

3) Newly-developed signal processing VLSIs:

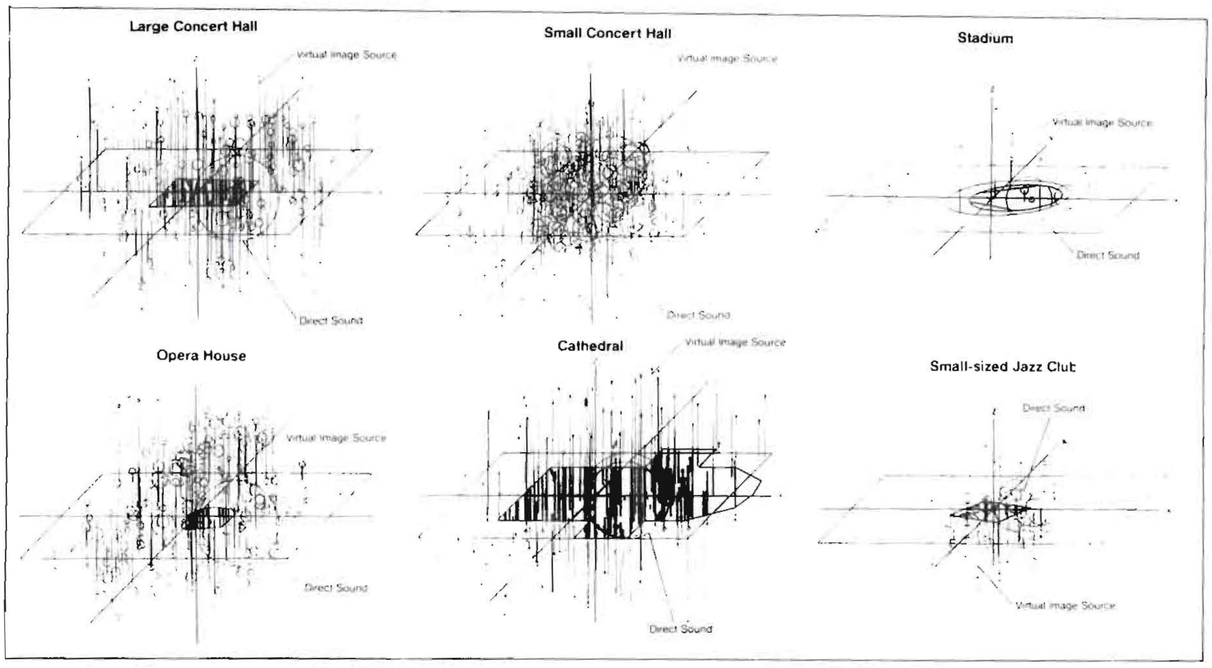
By considering all the above items, the signal processing data is all held in newly developed DAP VLSIs, including the data held in ROM memory chips which store the measurements made in the various concert halls, etc.

At the same time, unlike the other manufacturer's monaural processing system, our DAP system employs stereo (2-channel) signal processing, made possible by our 3-point source signal measurement.

Sound Field Reproduction by JVC's DAP System



Symmetrical 6-Point Sound Field Analysis Pattern



Acoustics Processing Mode Patterns

ACOUSTICS PROCESSING MODE PATTERNS

NO	PROGRAM NAME	TYPE	PARAMETER NAME	LOW	PRESET VALUE	HIGH
1	SYNPHONY HALL 1	SHOEBOX TYPE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	7 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.7	1
2	SYNPHONY HALL 2	SHOEBOX TYPE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	7 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.7	1
3	SYNPHONY HALL 3	SHOEBOX TYPE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	7 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.7	1
4	SYNPHONY HALL 4	VINEYARD TYPE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	7 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.7	1
5	SYNPHONY HALL 5	VINEYARD TYPE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	7 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.7	1
6	SYNPHONY HALL 6	VINEYARD TYPE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	7 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.7	1
7	RECITAL HALL	SMALL MUSICAL SPACE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	8 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.8	1
8	OPERA HOUSE	WITH TIERED SEATING	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	8 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.8	1
9	CATHE-DRAL	GOTHIC STYLE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	5 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.5	1
10	CHURCH	HIGH-CEILINGED SPACE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	6 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.6	1

NO	PROGRAM NAME	TYPE	PARAMETER NAME	LOW	PRESET VALUE	HIGH
11	LIVE CLUB 1	JAZZ CLUB	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	16 kHz	THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	1	—
12	LIVE CLUB 2	DISCO-THEQUE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	16 kHz	THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	1	—
13	PAVILION	LIVE CONCERT	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	6 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.6	1
14	GYMNA-SIUM	HARD-FLOORED HALL	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	8 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.8	1
15	STADIUM	OUTDOOR LIVE CONCERT	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	3 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.3	1
16	MOVIE THEATER 1	SMALL SPACE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	16 kHz	THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	1	—
			REAR DELAY	15 ms	20 ms	30 ms
17	MOVIE THEATER 2	MEDIUM-SIZED SPACE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	8 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.8	1
			REAR DELAY	15 ms	20 ms	30 ms
18	MOVIE THEATER 3	LARGE SPACE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	7 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.7	1
			REAR DELAY	15 ms	20 ms	30 ms
19	MOVIE THEATER 4	EXTRA LARGE SPACE	ROOM SIZE	0.5	1	2
			LIVENESS	0.5	1	2
			LPF	1 kHz	8 kHz	16 kHz, THRU
			REVERB LEVEL	0	1	2
			HF REVERB	0.1	0.8	1
			REAR DELAY	15 ms	20 ms	30 ms
20	MOVIE THEATER 5	STAND-ARD	REAR DELAY	15 ms	20 ms	30 ms

Table of Acoustic Processing Mode Patterns

HALL 1: Typical European concert hall with about 2200 seats.
HALL 2: Large concert hall with about 2500 seats; with dynamic acoustic characteristics.
HALL 3: Multi-purpose hall with about 1000 seats.
HALL 4: Large concert hall with about 2500 seats surrounding the stage.
HALL 5: Large hall with diffraction characteristics such that direct sound and reflections are well-balanced.
HALL 6: Multi-purpose hall with about 1500 seats, with "dead" ambience.
MOVIE THEATER 1: A small, cubic movie theater with about 100 seats.
MOVIE THEATER 2: Normal movie theater with about 400 to 600 seats.
MOVIE THEATER 3: Medium-sized movie theater with about 1000 seats.
MOVIE THEATER 4: Large movie theater with about 1500 seats.
MOVIE THEATER 5: Suitable to give a surround sound effect, similar to Dolby Surround sound, to movie programs.

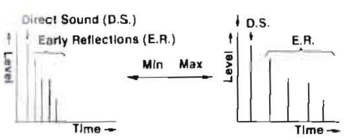
Parameters

	Parameter	Adjustable Range	Step	Initial Value	Remarks
1	ROOM SIZE	0.5 — 2	0.1	1	Can be stored in Manual Preset Memory
2	LIVENESS	0.5 — 2	0.1	1	
3	LOW PASS FILTER	1 — 16 kHz, THRU	1 kHz		
4	REVERB LEVEL	0 — 2	0.1	1	
5	HIGH-FREQ REVERB	0.1 — 1	0.1		
6	OFFSET DELAY	0 — 200 ms	1 ms	0	
7	REAR DELAY	15 — 30 ms	1 ms	20 ms	

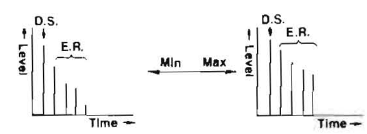
	Parameter	Adjustable Range	Step	Initial Value	Remarks
8	SPREAD/POINT	SPREAD/POINT		SPREAD	Last setting held in Memory
9	LISTENING ROOM REVERB	0.2 — 0.6 sec	0.1 sec	0.4 sec	
10	LISTENING ROOM SIZE	10 m² or less, 10 — 16 m², 16 m² or more		10 — 16 m²	
11	SOURCE REVERB	0 — 5 sec	0.1 sec	0 sec	

The 20 preset patterns consist of the parameters required by the DAP system to recreate the sound field in each of the places where music is performed. They are based on sound-field data measured using the symmetrical 6-point sound-field analysis method. You can create your own original sound field by changing one or more of these parameters, you can create an original sound field of your very own:

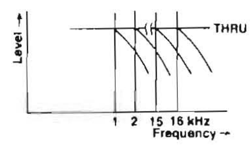
1. **ROOM SIZE**
The data on the size of the auditorium is programmed with a standard value of 1.0. You can change this parameter from 0.5 to 2, for an auditorium of half to twice the preset size.



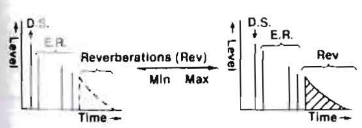
2. **LIVENESS**
The amount of sound reflected measured in the auditorium is programmed with a standard value of 1.0. You can change this parameter between 0.5 (half) and 2 (double).



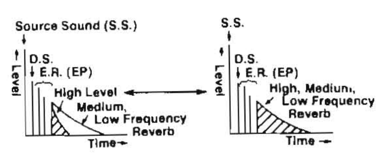
3. **LPF (Low-Pass Filter)**
Indicates the overall frequency characteristics of the reflected sound. The programmed standard value varies depending upon the auditorium. You can select cut-off frequencies of 1 kHz to 16 kHz or by-pass the low-pass filter.



4. **REVERB LEVEL**
Controls the reverberation sound level. The data of each performance site is programmed with a standard value of 1.0. You can change this parameter between 0 (no reverberations) and 2 (twice as much reverberation).



5. **HF REVERB**
Controls the reverberation time characteristics at high frequencies. The standard value of 1.0 gives the same characteristics as at middle frequencies. You can change this parameter between 0.1 and 1.0 (double). The standard value varies depending upon auditorium where the data was measured.



6. **REAR DELAY (Preset 16 — 20 only)**
Sets the time by which the R/DAP output signals (the rear speakers) are delayed relative to the main signals. You can change this parameter from the standard value of 20 msec to between 15 and 30 msec.

7. **OFFSET DELAY**
Sets the delay between the signals supplied to the F/DAP (front speakers) and the R/DAP (rear speakers) and the signal driving the main speakers in order to compensate for the time delay between digital input signals and analog output signals of an amplifier with a built-in D/A converter. You can change this parameter from the standard value of 0, up to a maximum of 200 msec.

Notes:
i) With analog input signals, you do not need to use this parameter.
ii) The R/DAP output of preset patterns No. 16 — 19 cannot be compensated by this parameter.

■ Tips for the effective demonstration of the XP-A1010TN

Together with the XP-A1010TN, in addition to an existing 2-channel stereo system, two additional pairs of DAP (sound field) speakers and two amplifiers are required for the 6-channel sound field reproduction, or one additional pair of speakers and an amplifier, for 4-channel system.

As shown in Fig. 7, these DAP (sound field) speakers should be placed at the four corners of the listening room, surrounding the listener.

With this speaker arrangement, you can appreciate the resultant effect, however, if the DAP (sound field) speakers are placed more precisely, a more accurate and realistic effect will be obtained. The following shows how to obtain the optimum effect.

As shown in Fig. 8-(B), the shape of the listening room is typically rectangular. (Here the length-to-breadth ratio is 1.5 : 1.)

The two speakers for the front-DAP channels should be located at a distance 1/10 · A (distance from front to rear of the room) behind the main 2-channel speakers.

The two speakers for the rear-DAP channels should be located at points 1/3 · A behind the listener.

These four DAP speakers should be placed slightly higher than the level of the listener's ears, as shown in Fig. 8-(A).

The values in these diagrams can be used as a standard so long as the room is rectangular. Fig. 8 shows the average setting condition of the sound fields measured in various listening rooms.

You might think that the main speakers are too close together. This is because the main speakers are used to reproduce the normal stereo sound as the direct sound from the sound source, while the front and rear DAP (sound field) speakers are used to reproduce the reflections and reverberations. Therefore, the front main speakers should be placed closer together than in normal stereo reproduction.

When, however, the XP-A1010TN is used in the 4-channel mode, the main speakers should be placed slightly further apart. This is because, in this mode, the main speakers are used not only to reproduce direct sound but also for the reproduction of the front-channel sound field signals.

(Although this 4-channel configuration is possible, it is recommended that the XP-A1010TN is operated in its "normal" 6-channel mode to obtain full benefit from it.)

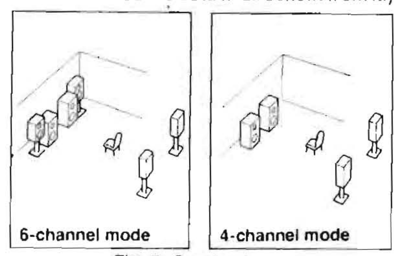


Fig. 7: Speaker Layout

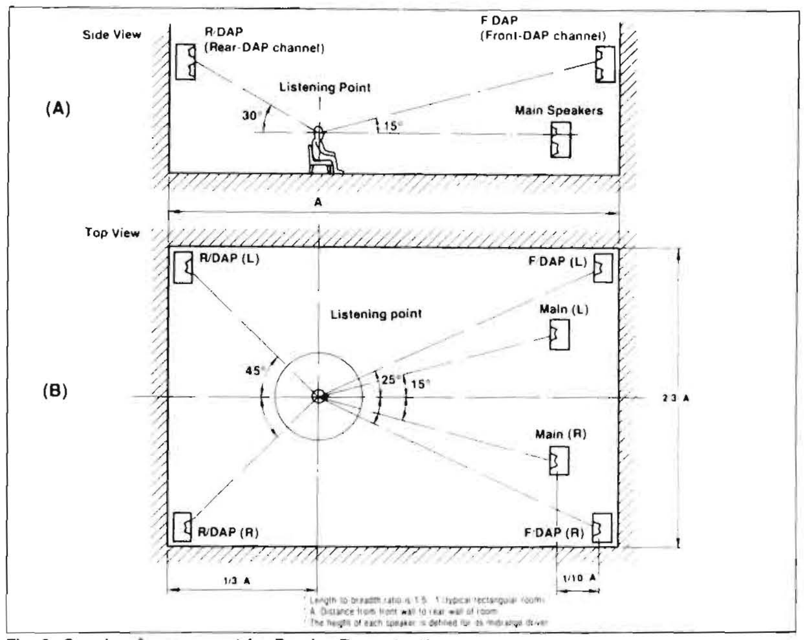


Fig. 8: Speaker Arrangement for Precise Reproduction

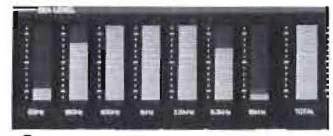
■ The following table shows source material we recommend for effective demonstrations

	Genre	Tune Title (Example)	Disc Title (Composer)	Sound Field Pattern	Effective demo method
1	Classic	"Light Cavalry"	Suppe	No. 1 (Symphony Hall 1)	2-CH → Preset No. 1
2	Jazz	"High Jingo"		No. 11 (Live Club 1)	2-CH → Preset No. 11
3	Popular	"Help"	Beatles Collection	No. 7 (Recital Hall)	2-CH → Preset No. 7
4	Big Band	"Keeping Count"	Tokyo Union	No. 13 (Pavilion), No. 15 (Stadium)	2-CH → No. 13 → 2-CH → No. 15
5	Classic (Organ)	Symphony No.3, 4th Movement	Saint-Saëns	No. 9 (Cathedral)	2-CH → Preset No. 9
6	Opera (Air)	"Toreador's Song" from "Carmen"	Bizet	No. 8 (With tiered seating)	2-CH → Preset No. 8
7	POINT/SPREAD comparison	"Eine kleine Nachtmusik"	Mozart	No. 5 (Symphony Hall 5)	POINT ⟶ SPREAD
8	Hall type comparison	(Same as above)		No. 1 (Symphony Hall 1) No. 4 (Symphony Hall 4)	No. 1 ⟶ No. 4 Compare "shoebox" and "vineyard" concert halls.
9	AV source	Dolby Surround-encoded video source	Sci-fi movies, etc. are especially suitable.	No. 16 ~ 19 (Movie Theater 1 ~ 4), No. 20 (Standard)	Compare theaters of different sizes. (No. 20: Same effect as normal Dolby Surround)

S.E.A. GRAPHIC EQUALIZER

Feature Highlights of '90 S.E.A. Graphic Equalizer

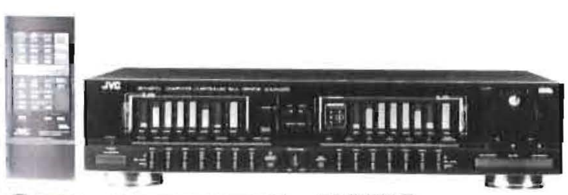
1 *7 electronic frequency controls for each channel, plus real-time spectrum analysis*



2 *6 preset and 6 user-programmable equalization patterns*

3 *Full-function remote control with motor-driven volume control*

Computer-controlled 7-band S.E.A. graphic equalizer with remote control



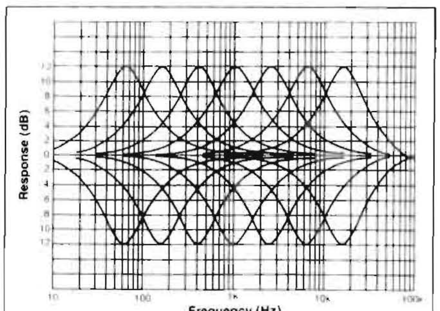
SEA-M770BK DIGIFINE

S.E.A. Graphic Equalizer

- 7 electronic frequency controls for each channel
- Full-function remote control with motor-driven volume control
- 6 preset and 6 user-programmable equalization patterns
- Real-time spectrum analysis
- Transfer function

Feature Comparison Chart

	SEA-M770BK
Control	
Electronic Control	✓
Control Elements	7 + 7
Remote Control Unit	✓
Electro Servo Volume	✓
Synchro Control	✓
Display	
Spectro Peak Indicator	✓ (Peak Hold)
Display Level	✓
Memory	
Number of User Program Memory	6
Number of Programmed Memory	6
Function	
Source	✓
Rec	✓
Tape Monitor	✓
Preset Scan	✓
Transfer Function	✓



S.E.A. Characteristics

Transistor inductors (SEA-M770BK)

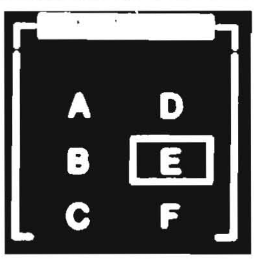
JVC equalizers use semiconductor inductor elements consisting of a transistor, a resistor and a capacitor in an original JVC circuit construction in the SEA resonance circuits for higher sound quality than can be obtained using coil inductors.

Semiconductor inductors are highly resistant to induction noise and magnetic distortion that can occur with the coils used for filtering in conventional graphic equalizers. They result in sharp filtering characteristics and make smooth operations possible. With JVC's exclusively-designed circuit, equalization will add nothing to degrade the source signal being compensated; this is particularly important when equalizing signals from a compact disc or other digital source which has little inherent deterioration of sound quality.

Electronic Up/Down level controls (SEA-M770BK)

Instead of mechanical levers and variable resistors, the SEA-M770BK uses an LSI containing electronic switches. This LSI chip developed exclusively for S.E.A. contains about 3800 elements and permits all functions to be operated both from the main unit and the remote control. Each frequency band is controlled electronically, with UP and DOWN switches provided to boost and attenuate response.

Equalization has never been easier. Simply touching the UP or DOWN button will boost or attenuate the desired frequency band. The equalized or compensated result can be checked at a glance referring to the FL bar display for correct and optimum compensation. With their purely electronic operation, these switches provide trouble-free convenience and simple operation as well as high tracking precision.

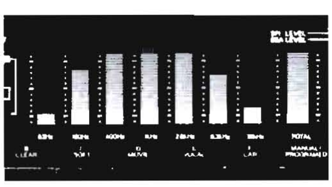


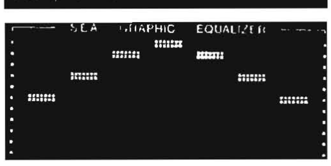
6 programmable preset patterns (SEA-M770BK)

In the SEA-M770BK, 6 "PROGRAMMED" preset equalization patterns are provided, and can be recalled at any time. The 6 preset patterns are for "Heavy", "Clear", "Soft" "Movie", "Vocal" and "Car" equalization. In addition, the same number of user-programmable "MANUAL" patterns are also available.

The user can select the desired equalization pattern instantly by simply touching the corresponding button. For example, if the user likes hard rock music, press the "Heavy" button to recall the optimum equalization pattern for pop and rock music. Or if he or she likes light background music, the "Soft" response will be most suitable. Furthermore, the user-programmable memory accepts up to 6 equalization patterns for the more critical, personalized requirements of users. The user can find the optimum response in his/her listening room or set up any customized characteristics to be stored in memory for different sources, etc., to be recalled at any time by just touching a button.

Refer to "Demonstration" page 79 for more information.



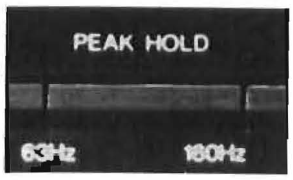


SEA/SPI FL display (SEA-M770BK)

The SEA-M770BK has S.E.A./S.P.I (Spectro Peak Indicator) FL displays for each channel. When performing compensation, the FL display shows the response in each frequency band; in the SPI mode, it becomes a real time spectrum analyzer that shows the output level of each band with the varying input signal. Display level controls allow the range of the display to be adjusted for high and low level signals.

The frequency response display helps the user to "visualize" the music and compensate for room acoustics and frequency characteristics to suit his or her requirements. Equalization can be done easily while watching the FL display. The user can switch the display to a real-time spectrum analyzer by simply pressing a button. Demonstrating the SPI display moving up and down according to the level of the music signal will attract the interest of potential buyers in a shop-front display.

Technology/Function	Benefits



Peak hold button (SEA-M770BK)

The SEA-M770BK has a Peak Hold function used in the SPI display. When this is activated, the peak levels of each frequency are held until the next higher value is reached.

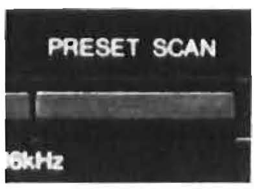
In the SPI mode with the SEA-M770BK, the Peak Hold button allows the peak levels to be indicated for several seconds in each frequency band. This function will be very helpful when checking the peak components of the source signal for recording, etc.

Remote-controlled motor-driven volume control (SEA-M770BK)

In the SEA-M770BK, the output volume is controlled by a built-in motor which can also be controlled with the provided remote control unit.

Since the SEA-M770BK accepts the rec out signal or pre out signal from the amplifier, the overall volume level of any source signal (such as from the CD player or cassette deck) selected by the amplifier's function switch can be controlled via the graphic equalizer. The motor-driven volume control which incorporates an indicator permits the user to adjust the volume from a distance with the remote control unit while watching the easy-to-check LED indication. This does not cause signal deterioration because the volume control is driven by a special motor.

Refer to "Demonstration" page 79 for more information.



Preset scanning of preset patterns (SEA-M770BK)

In the SEA-M770BK, each of the 6 programmed-preset characteristic patterns and 6 user-programmable preset patterns can be recalled in sequence, by pressing the Preset Scan button.

With this function, the user can recall each preset pattern by simply pressing a button for an easy and convenient check of the results of each setting. Each of the programmed and user preset patterns will be recalled sequentially. When the best setting is found, pressing the Preset Scan button again holds it. These operations can also be performed from a distance with the provided remote control unit, while checking the results from your listening position. The user can easily search for the best or most appropriate setting to suit the source and mood.

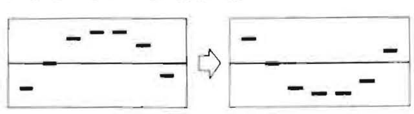
Response transfer function (SEA-M770BK)

With the SEA-M770BK, the compensated response for one channel can be transferred to the other channel. Entirely the same equalization effect of one channel can easily be applied to the other channel.

This helps the user to compensate the response of both channels to obtain the same equalization effect without performing the same control operations twice. Transferring can be done from the left channel to the right channel or vice versa by pressing the transfer direction button. If fine adjustment is required, the user can minutely adjust the required channel settings after transferring the response from one channel to the other.

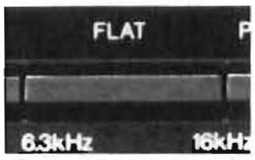
Characteristics reversal (SEA-M770BK)

This function reverses the compensated response characteristics after equalization. That is, frequencies boosted in recording will be attenuated by the same amount in playback, while frequencies that have been attenuated will be boosted.



To use this function, the high frequencies boosted in recording will be attenuated by the same amount when the recorded tape is played back, thereby achieving the same effect as a noise reduction circuit. When frequency response data for the speaker system is available, a flat speaker frequency response can be obtained instantaneously, by pressing the Reverse button.

Technology/Function **Benefits**



Flat response button (SEA-M770BK)

JVC's SEA graphic equalizers have a Flat button for the instant recall of a flat frequency response.

With this function, flat response can be recalled instantaneously at any time, even during equalization. That is, when adjusting the equalizer knobs or switches, pressing the Flat button will instantaneously recall flat amplification of the source signal. This function will be very helpful when comparing the equalized results with the original, etc.

Synchro control (SEA-M770BK)

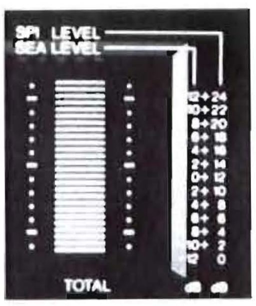
In the SEA-M770BK, with the Synchro button ON, level up/down adjustments made for one channel will be performed simultaneously for the other channel.

This function enables the user to operate the L and R channel control buttons simultaneously. This will be very helpful when the same compensation is required for both channels.

Fade muting on remote control unit (SEA-M770BK)

The Fade Muting button on the remote control unit provided with the SEA-M770BK allows the output volume level to be faded gradually at a fixed rate of attenuation.

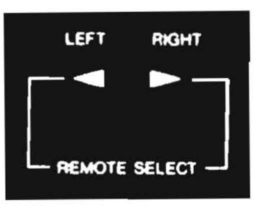
The output volume level can be attenuated instantaneously from a distance with the remote control unit, but the attenuation will be performed gradually at a fixed rate, with a smooth fading operation instead of the sudden muting usually provided with an amplifier. The reproduced music will fade away more naturally.



±12 dB control range (SEA-M770BK)

The SEA-M770BK provides a control range of ±12dB for each frequency band so that the level of any frequency can be raised or lowered by any required amount.

The wide ±12 dB control range allows the user to compensate or equalize the required frequency band precisely and correctly. Fine adjustment of each frequency band will make the resultant characteristics optimum for any desired response. The user has fine control over response in the different bands as the SEA-M770BK provides a sharp and solid equalization effect in each frequency band, while the wide control range adds flexibility.



Remote selector (SEA-M770BK)

The SEA-M770BK has an exclusive remote control unit; most of its keys are the same as the front panel controls, for adjustment from a distance. The Remote Selector on the remote control unit also selects whether the L, R or L + R source signal is to be equalized, while the front panel Remote Select indicator shows the user's selection.

The user can select the input source to be equalized with the Remote Selector button on the remote control unit. Each time the Remote Selector button is pressed, the left, right or both channels of the source are selected in sequence; the selection can be easily checked from the Remote Select indicator on the front panel of the main unit. A pair of arrows lights in green to show that which signal is selected. This indicator is highly visible; bright against a solid black panel, the user can see it at a glance, even from a distance.

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Demonstration

SEA-M770BK

Employing advanced microcomputer technology. the SEA-M770BK is a fully-computerized multi-function graphic equalizer, with a full-feature remote control unit.

Probably a demonstration using the remote control unit will be most attractive, as follows.

- **SEA display on the FL panel** — Show it while varying the levels of each frequency band with the remote controller.
- Also demonstrate the **character transfer function**, from the L-ch to the R-ch, and vice versa.
- The **spectrum analyzer (SPI) display** will be especially attractive when placed in the store-front — Input the source (such as music from a CD player) and display the audio spectrum of the output sound. At this time, adjust the display level so that it is most noticeable.
- The **motor-driven remote volume** — one of the most attractive points. Hand the

remote controller to interested customers to let them operate it while watching the Volume knob on the front panel which moves as the volume is changed. The moving red indicator is an eyecatcher.

The **fade muting** key allows the volume level to be attenuated gradually. And more, since the SEA-M770BK has a switched AC outlet on its rear panel, when the amplifier's power plug is connected to this, the remote controller allows the user to **switch the entire hi-fi system on and off**.

This extraordinarily convenient feature will be a powerful selling point for a customer who owns a CD player or stereo system without a remote volume control. With the SEA-M770BK added to the system, increasing or decreasing the overall volume, and muting the sound, as well as turning the power of the whole system on and off can be done from the remote control.

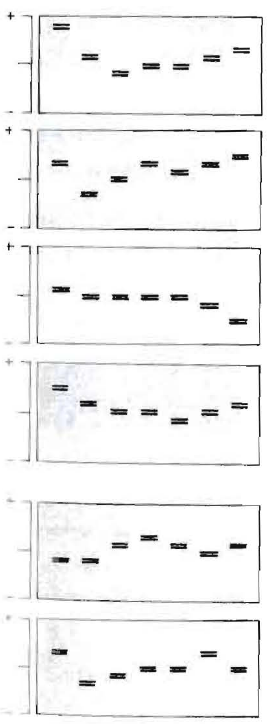
So, in demonstrating the SEA-M770BK, these functions will effectively show how total control of the volume is possible from a distance.

- **6 fine-adjusted "programmed" patterns** — recall each pattern while changing the input source (with the amplifier's selector). Be sure to explain the characteristics of each pattern referring to the description below.
- **6 user-programmable "manual" patterns** can be stored — first recall any of the "programmed" patterns, and modify it as required, then store the modified pattern in any of the "manual" memories. After this, you can recall it to demonstrate how easy and convenient it is to store and recall equalization patterns.

Six "programmed" equalization patterns

With the SEA-M770BK, the following 6 equalization patterns are preset during the manufacturing process. Any of these can be recalled by simply touching a button.

Use them depending on the situation or the customer's taste in music when performing a demonstration; refer to the brief guide below for a description of each pattern.



HEAVY
Ideal for music with a heavy beat such as rock. With this setting, low frequencies are emphasized to produce sound that's deep and powerful. Higher frequencies are also boosted to add clarity to percussion and other instruments.

CLEAR
This setting provides crisp and clear sound with soaring highs. Lows and mid frequencies that tend to be muffled are toned down. The mid to high frequencies are emphasized to bring the vocals into prominence.

SOFT
Suitable for background music or easy listening music that doesn't sound harsh. Deep low frequencies that tend to be lost at lower levels are given more emphasis. Highs are subdued to achieve better overall balance.

MOVIE
This setting is perfect for video sound — from TV, video tapes or videodiscs. With video programs, lows and highs tend to be weak, so they are given a boost to produce balanced overall sound, while most other frequencies are reduced. Since video programs sometimes sound too bright, mid-frequencies are cut.

VOCAL
For singing or narration. This position puts emphasis on the mid-range "vocal" frequencies to make voices more distinct, while reducing other frequencies. To accent the higher vocal registers, high frequencies are also boosted.

CAR
For making tapes to be played in a car. As you may know, the acoustics of each car's interior are very different for each model. However, in most cases, middle-low frequency sound tends to be muffled due to the restricted space and, with speakers installed at the rear, high frequencies are attenuated by the time the sound reaches the driver. To counter this, mid-low frequencies are attenuated while high frequencies and very low frequencies are boosted.

AV SURROUND PROCESSORS

Feature Highlights of '90 AV Surround Processors

1

DOLBY Surround Sound circuitry takes full advantage of music videos encoded with three-dimensional Dolby Stereo, for total entertainment

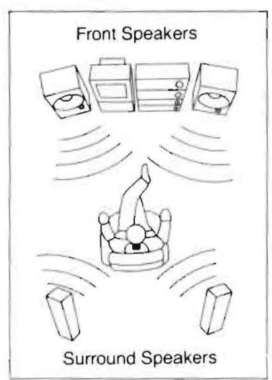
DOLBY SURROUND

2

Remote control capability with master volume control

3

Hall Surround and Simulated Surround modes expand conventional stereo and mono sources



Feature Comparison Chart

	SU-A400	SU-A30
Surround Mode		
Dolby Surround	✓	✓
Hall Surround	✓	✓
Simulated Surround	✓	✓
Circuit		
Built-in Rear-Channel Amp	✓	✓
Surround Headphone Jack	✓	✓
Pre-Out (Surround) Terminals	✓	✓
Tape Monitor Terminals	✓	
Function		
L/R Calibration	✓	✓
Delay Time Control	✓	✓
Variable Effect Level	✓	✓
Peak Level Indicator	✓	✓
Master Volume	✓	
Calibration Level Meter	✓	✓
Remote Control (Provided)	✓	

Lineup of '90 AV Surround Processors

A/V surround processor with remote control

- Dolby Surround, Hall Surround and Simulated Surround
- Wireless remote control
- Master volume control with indicator
- Built-in 10W + 10W power amplifier
- L/R calibration and delay time control

SU-A400

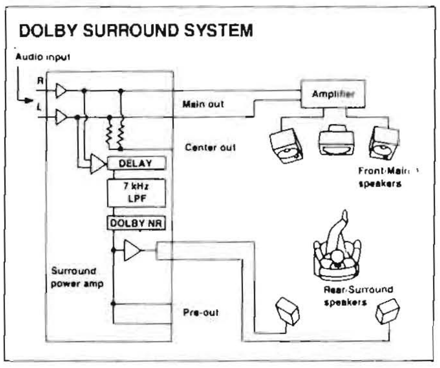
A/V Surround Processor

A/V surround processor with 10W + 10W power amplifier

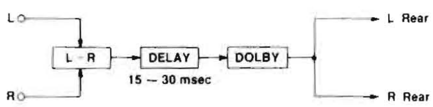
- Dolby Surround, Hall Surround and Simulated Surround
- L/R calibration and delay time controls
- Rear-channel volume control
- Built-in 10W + 10W power amplifier

SU-A30

A/V Surround Processor



* Main Audio Out and Center Out jacks are not provided
DOLBY SURROUND circuit in the SU-A400/SU-A30.



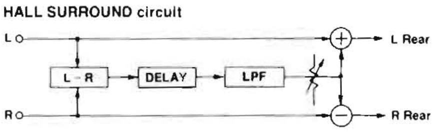
* DOLBY SURROUND and the double-D symbols are trademarks of Dolby Laboratories Licensing Corporation

Dolby Surround decoder (All models)

The SU-A400/SU-A30 incorporates a Dolby Surround decoder. In the processing circuit, the left-right difference signal is delayed and filtered with Dolby NR and output monaurally from the rear speakers. For correct calibration, independent L/R Calibration controls and Peak Level Meters are provided. The delay time can also be controlled between 15 and 30 msec with the independent Delay Time control.

Many movies have soundtracks encoded with the Dolby Surround system. With a Dolby Surround decoder, these movies can be reproduced with a 3-dimensional effect similar to that experienced in a movie theater.
Because independent L/R calibration controls are provided, the user can adjust the response to obtain the optimum effect while checking the LED level meters. For an even better effect, so that the reverberation time can be set to the ideal value for different listening/viewing rooms, an independent delay time control is provided.

HALL SURROUND circuit

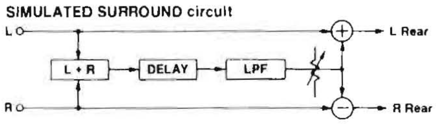


Hall Surround (All models)

In this JVC's exclusively-designed circuit, the left-right difference signal is delayed and filtered by an LPF to be output to the rear L and R channels. The characteristics of the filter have been carefully determined by JVC engineers. With the Hall Surround mode, the sound field from a normal stereo source is expanded to simulate a large concert hall.

For movies and music video programs which are not encoded with the Dolby Surround effect. A 3-dimensional effect can be obtained even when the sound track is not Dolby Surround-encoded in this mode; especially suitable for films of concerts and conventional stereo sources. The concert hall effect is especially desirable when a film of a live concert (classical music or rock) is used. The delay circuit helps to add depth and reverberation to the sound.

SIMULATED SURROUND circuit



Simulated Surround (All models)

With the Simulated Surround mode, the mono signals are processed to create a quasi-stereo sound field. The rear speakers output the same monaural sound but it is delayed to provide an expanded sound field. In this circuit, the monaural L + R signals are delayed and filtered with the LPF providing the optimum effect.

With the Simulated Surround mode, monaural sources, which make up a greater part of conventional video and TV programs, can also be enjoyed with the surround effect. This mode also uses the delay circuit to apply width and depth to the sound field which can be optimized with the Delay Time knob.

Remote control capability (SU-A400)

The AV Surround Processor SU-A400 comes with an infrared wireless remote control unit, which allows the user to control the important functions from a distance. It controls the selection of the surround mode, activates tape monitoring, controls overall volume, and also switches the muting function on and off for added convenience.

With remote control capability, the user can select the required surround mode from his/her listening position, while adjusting the overall volume of the front and rear channels. And if SU-A400 is used together with a JVC remote controlled receiver, it can be controlled with the single remote control unit provided with the receiver (instead of using two different remote controllers).

Master volume control (SU-A400)

The SU-A400 provides a master volume control which allows the control of the overall volume level for the front and rear speakers simultaneously, while the independent rear volume level control control of the rear-channel speaker level; with this, it can be used for front-rear balancing. Volume control buttons with the same functions are also provided on the remote control unit.

After adjusting the balance between the front and rear channel signals, the master volume gives overall control for the adjustment of the sound level from all speakers simultaneously. Since the provided remote control unit also has the same master volume control buttons, the overall level of the four speakers can be adjusted from your listening position without affecting the front-rear balance. You don't have to adjust both the front-channel amplifier's volume control and the rear-channel volume control separately.

Built-in power amplifier with volume control (All models)

An amplifier with an output of 10 watts per channel is incorporated in the SU-A400/ SU-A30 so it can drive rear/surround speakers directly.

With JVC's Surround Processor, the user can enjoy the surround sound by simply adding rear/surround speakers (even old-fashioned or small speakers will do). The 10 watts + 10 watts power amplifier will drive these rear/surround speakers with enough power to produce an excellent surround sound effect. The level of the sound from the surround/rear speakers relative to that of the front speakers can be adjusted to balance between the front and rear speakers.

Pre-out (surround) terminals (All models)

The SU-A400 and SU-A30 are also equipped with pre-amplifier (surround) output terminals for the connection of an external power amplifier.

The user can upgrade the sound quality of the rear/surround speakers by connecting an external power amplifier to these terminals. These terminals output the decoded signals necessary to produce the surround effect which can be amplified independently by an external power amplifier, resulting in surround sound with greater dynamic.

Surround Sound Headphones jack (All models)

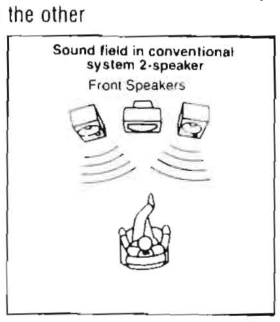
The SU-A400/SU-A30 has a headphone jack which will accept the specially-designed Surround Sound Headphones.

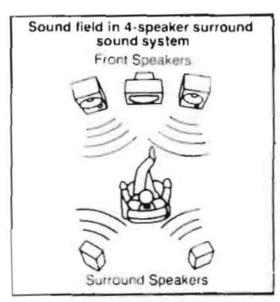
A unique feature of JVC's Surround Processors is their Surround Sound Headphone jack which outputs the surround signals, not just the main/front signals. When the optional Surround Sound Headphones are used, the user can enjoy the full potential of surround sound by him or herself without disturbing other people.

What is Surround Sound Effect?

In a live listening situation, whether classical music in a concert hall, live rock music in a club or in the movie theater, sound is heard not only from the front but there are also echoes from the sides, rear walls and ceiling, creating a "multi-dimensional" sound field. However, the normal 2-speaker stereo system only creates a two-dimensional sound field with sound from the front. On the other hand, with rear surround speakers added, reflections and reverberations created by the delay and specially-designed signal processing circuitry, are output from these rear/surround speakers. This greatly expands the sound field when compared to sound from the front speakers on their own.


Sound field in live performance or in the movie theater


Sound field in conventional system 2-speaker


Sound field in 4-speaker surround sound system

Feature Highlights of '90 Programmable Remote Control

1 *Extremely flexible programming functions let you replace other infrared remote controls*

2 *Touch-sensitive LCD panel with menu screens for 11 different source components*

3 *Standard mode for control over JVC AV components plus programmable mode for control over other components*

Universal AV programmable remote control with touch sensitive LCD panel screen

RM-S1

Universal Programmable Remote Control Unit

- Programmable with up to 200 commands of other remote controls (with JVC standard code length)
- Touch-sensitive LCD panel
- Menu screens for 11 different sources
- Variable-length coding system
- Two display modes: JVC standard with 10 menus and programmable with 18 menus

Technical Notes

■ Truly unified home remote control

A single remote control unit that controls not only JVC remote-controllable audio/video components, but that can be programmed with the codes of other manufacturer's infra-red remote controls.

Many recent home entertainment components come with remote controls, extending from video components like TV sets and VCRs, to audio components like CD players and amplifiers, and even to home appliances such as air conditioners.

JVC's RM-S1 can learn all of these remote control codes and hold them in its memory, so they can be operated from a single remote control.

■ Full-programmable functions

The RM-S1 is equipped with the touch-sensitive LCD panel screen which changes according to the equipment to be operated. When the user specifies the component to be remote controlled, by pressing the selector button, the LCD panel will change accordingly. In this way, the RM-S1 offers a maximum of 200 remote control functions (if all the codes are the same length as the JVC standard code).

There are screen menus for 11 program sources — two VCRs, a videodisc player, a TV, a CD player, a DAT deck, a conventional cassette deck, a tuner, a turntable, an auxiliary component (AUX), and a utility unit (EXT). In addition, the amplifier or receiver used as the A/V system's control center can also be controlled with the RM-S1's "programmable" buttons.

To achieve this flexibility, the LCD panel has the two display modes; a [JVC standard] mode with 10 menu screens, and a [Programmed] mode with 18 menu screens.

■ A JVC exclusive

Unlike the "learning" or "programmable" remote controls from other manufacturers, our RM-S1 employs a new variable-length coding system.

Because the infrared beam coding systems used by different manufacturer and types of A/V component are different, most programmable remote controls accept the longest possible remote control codes. However, if the length of the codes is shorter than this, only part of the memory is actually used to hold codes, and the rest will be wasted, which limits the number of programmable functions.

But with our innovative variable-length coding system, the memory area for storing the remote control codes can be varied according to the transmitted codes from the other remote controls. Therefore, the number of programmable codes is greatly expanded, while the wasted space are reduced.

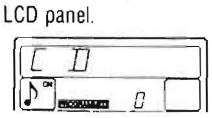
As a result, the RM-S1 can learn up to 200 remote control codes when the remote control codes to be programmed have the same code length as that of the JVC standard ones.

■ Programming procedure

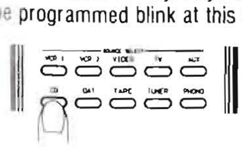
Programming other remote control codes is quite simple.
Perform in a following manner:

1) Set the USE/LEARN switch on back of the RM-S1 to "LEARN".

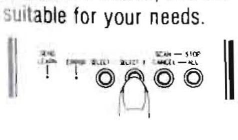
The [PROGRAMMED] indication appears in the LCD panel.



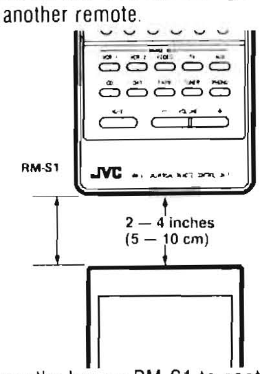
2) Call up the menu screen corresponding to the required source on the LCD using the SOURCE SELECT key: Keys which can be programmed blink at this time.



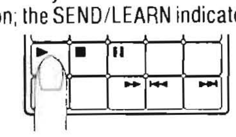
3) Using SELECT-2 button, find the menu most suitable for your needs.



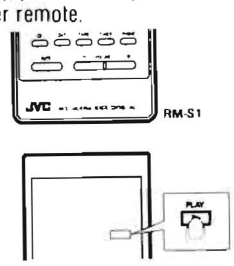
4) Place the RM-S1's end facing to the top of another remote.



5) Press the key on RM-S1 to contain the function; the SEND/LEARN indicator blinks.



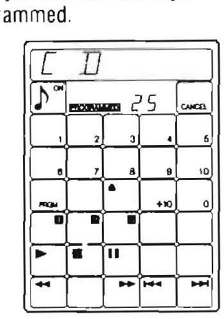
6) While the indicator is blinking (within 7 seconds), press the required button on the other remote.



7) Repeat the above steps to program other functions.

8) Set USE/LEARN switch back to "USE".

9) Only the programmed keys will be visible on the LCD panel.
Now, you can use the keys programmed.



TURNTABLES

Feature Highlights of '90 Turntables

1 ***Our COMPU LINK control system makes possible remote control and allows interactive operations with other components***

2 ***JVC's newly developed high-stability cabinet support system minimizes acoustic resonance***

3 ***International standard plug-in cartridge connectors make cartridge connection simple***

Feature Comparison Chart

		AL-FQ555BK	AL-F353BK	AL-A151BK
Type				
Drive System	**Direct Drive**	✓		
	Belt Drive		✓	✓
Quartz-Lock		✓		
Mechanism				
Motor	**Coreless DC FG Servo Motor**	✓		
Operation Mode	Fully Automatic	✓	✓	
	Auto Return			✓
Suspension System	New Cabinet Support System	✓		
Others				
Strobe		✓		
Cartridge Connector	T4P Plug-In Cartridge	✓	✓	✓
Auto Disc Size Selector		✓	✓	
Auto Speed Selector		✓	✓	
Tonearm Cueing Control		✓	✓	✓
Space-Efficient Dust Cover		✓	✓	✓
COMPU LINK Compnent		✓ (Play/Stop)	✓ (Play/Stop)	

Quartz-locked fully automatic direct-drive turntable with COMPU LINK control

AL-FQ555BK COMPU LINK Component

Turntable

- Double-servo quartz-locked control
- Coreless DC direct-drive motor with super FG servo
- Fully automatic operation
- Low-mass straight tonearm with low-center-of-gravity support
- New cabinet support system and large insulators

Fully-automatic belt-drive turntable with COMPU LINK control

AL-F353BK COMPU LINK Component

Turntable

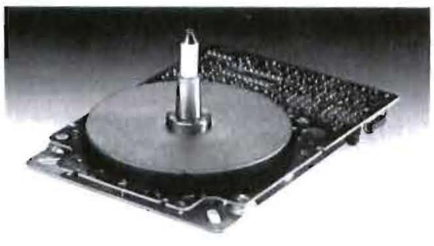
- Low-mass straight tonearm with low-center-of-gravity support
- Precision DC servo motor
- Fully automatic operation
- Plug-in cartridge connector
- Auto disc size/speed selector

Auto-return belt-drive turntable with low-mass straight tonearm

AL-A151BK

Turntable

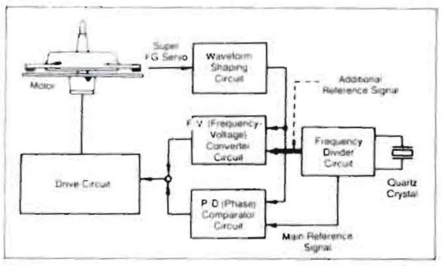
- Low-mass straight tonearm with low-center-of-gravity support
- DC servo belt-drive motor
- Plug-in cartridge connector
- Automatic arm return
- Arm cueing control



Coreless DC direct-drive motor with super FG servo (AL-FQ555BK)

JVC's DD models feature a coreless DC motor, for cogging-free smooth platter rotation which results from eliminating the iron core which could cause variations in the rotation of the spindle motor, as well as large-sized rotor magnets for superior stability. Furthermore, JVC uses an originally developed Super FG (frequency generator) servo control system in our DD models. This checks the motor speed with uncanny accuracy by sampling the hundreds of pulses generated by the frequency generator circuit incorporated in the highly sophisticated speed-detection system. This is the ideal motor system for DD turntables.

Since the spindle of DD turntables is a direct extension of the motor shaft, there is no transmission system requiring a complicated structure, with reduced mechanical wear because there is no belt or idler. Moreover, with their larger magnets, our DC-driven motors substantially eliminate unnecessary oscillations arising from variations in AC current. Also, since the motor rotates at ultra low speed, its accuracy can be controlled electronically and stabilized over a long periods of use. With JVC's turntables, the platter always rotates smoothly at the correct speed by the super FG servo system. The user never has to worry about the pitch of the music varying as it is being played.



Double-servo quartz-locked control (AL-FQ555BK)

Our top models incorporate a double-servo quartz-locked control system in addition to the super FG servo control system, for even more accurate speed control. In this circuit, the rotation of the platter (that is, the motor in a direct-drive system) is also monitored and corrected by a phase comparator and F-V converter circuit which are used to compare the signals generated by the rotation with a reference signal obtained by dividing the frequency of a built-in quartz crystal oscillator, which generates pulses with an incredibly accurate frequency.

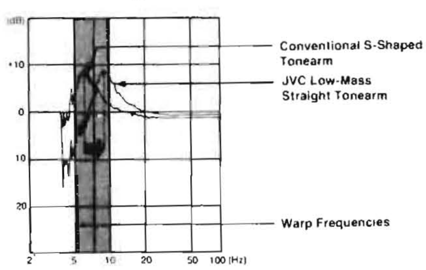
As a result, the stability of platter rotation is further improved, with greater resistance to the effects of variations in temperature, voltage, and load. This technology, in conjunction with the direct drive motor which drives the platter/record combination, ensures accurate speed even if the total weight of the platter and record is greater. Even when the load of the stylus on the record surface is high, as occurs in "highly modulated" passages, the speed is not affected, so there is no disturbance in the pitch of the music.

New cabinet support system with large insulators (AL-FQ555BK)

In the AL-FQ555BK, an anti-vibration structure called the New Cabinet Support System is used to overcome any unwanted acoustic feedback; large-sized insulators effectively support the turntable, rather than the conventional feet. In these models, the motor and tonearm section is isolated from the main chassis and its center of the gravity is shifted to a point above that of the cabinet, which is, in turn, supported by the large vibration insulators.

Vibrations and acoustic feedback from the speakers, etc. are effectively reduced before they reach the turntable cabinet, while our large-sized insulators have higher resistance to shocks as well as a stabilizing effect. This combination works together and results in higher resistance to disturbances caused by external vibrations or acoustic feedback. It allows the user to install the turntable in the position he or she wants, without having to worry about speaker/turntable placement.

Resonance-Damping Response of the JVC Low-Mass Straight Tonearm



Low-mass straight tonearm with low-center-of-gravity support (All models)

In tonearm design, JVC adopted the straight tonearm with lower mass to match the increasing number of high compliance cartridges, instead of the more usual S-shaped and J-shaped designs. Our low-mass straight tonearm has a resonance frequency region from approximately 6 Hz to about 10 Hz, while its center of gravity is positioned to be lower than that of most tonearms.

In practice, there are very few records without at least a little warping or eccentricity due to the vinyl material and mass-production presses. Playing these records can cause mistracking which creates audible performance loss and intermodulation distortion. However, our low-mass straight tonearm has a higher resonance frequency so they are not affected by the infrasonic frequencies generated by warped and eccentric records. At the same time, low center of gravity tonearm design reduces tracking distortion, improving "effective" wow & flutter and stability.

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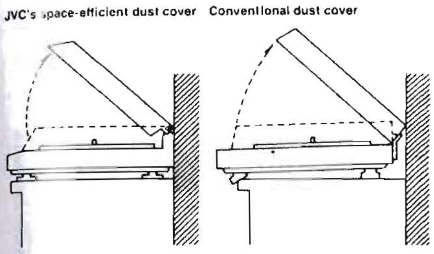
Illuminated strobe (AL-FQ555BK)

In the AL-FQ555BK, a strobe pattern is provided on the edge of the platter. This is illuminated by a built-in LED when the platter is rotating, and if the speed is correct, the stripe pattern will appear stationary.

During playing, the red LED lights to illuminate the stripe pattern; when this is stationary, it shows that the platter is rotating at the correct speed; 33-1/3 rpm for LP records or 45 rpm for singles. This shows at a glance that the speed is absolutely right. With the AL-FQ555BK, as the platter is driven by a quartz-locked DD motor, no adjustment is required.

JVC's space-efficient dust cover   Conventional dust cover



Space-efficient dust cover (All models)

Unlike most turntables which require a relatively large space behind them to allow the dust cover to swing open, JVC turntables have dust covers with a new design. The dust cover is hinged at the top rear of the deck, slightly above the platter, so it can be raised without protruding to the rear, whereas conventional dust covers are hinged at the rear of the cabinet.

Space-efficient dust covers do not require any clearance behind the rear panel of the cabinet. The user can install the turntable flush against the back wall without restricting the opening of the dust cover. This is a convenient space-saver which will make installation easier.

Plug-in cartridge connector (All models)

Turntables with the T4P mark are equipped with a plug-in cartridge connector. T4P is a standard which indicates that the components are "plug compatible" with each other.

This marks replacing cartridges easier as the user can choose a cartridge from many manufacturers to upgrade performance. Although the cartridge can be disconnected and connected easily, it is securely locked onto the tonearm. Mechanical strength, rigidity, and electrical contact are improved, while simple connection eliminates the chance of making wiring errors or poor contact.

Fully automatic operation (AL-FQ555BK, AL-F353BK)

Many JVC turntables feature fully-automatic operation, which allows all required operations to be performed automatically. Platter rotation and tonearm movement are accurately controlled by the precise mechanism which incorporates a microcomputer. Repeat play is also incorporated in these models for added convenience.

The user can play records simply by pressing one button without having to make adjustment or manually position the tonearm. The stylus moves over the outer edge of the record and is lowered onto the record surface when the platter starts rotating, and record play begins. When the record is finished or when the stop button is pressed, the tonearm is lifted up and returned to the arm rest and the platter stops. All these operations are performed automatically. The user doesn't have to worry about needless stylus and record wear due to continuous playing of the lead-out groove, even if he or she falls asleep before the record finishes.

Automatic disc size/speed selector (AL-FQ555BK, AL-F353BK)

With the AL-FQ555BK and AL-F353BK, the disc size is automatically selected by a built-in photosensor, making it easy and convenient to play records.

Most currently available records are divided into two types: 12", 33-1/3 rpm LPs and 7", 45 rpm singles. Therefore, when the photosensor detects the record size, the record speed is automatically set to the corresponding value. If you have records that don't conform to these standards, don't worry as our fully-automatic models also have a manual control position and non-standard records can be played manually.

SPEAKERS

Feature Highlights of SX-911WD Speakers

1 *"Fine" cloth-carbon midrange unit for enriched, natural mid-band response*

2 *Cloth-carbon woofer for tight and heavy bass response*

3 *Amorphous-diamond coated tweeter for crisp and clear highs*

Feature Comparison Chart

	SX-911WD	SX-A3
Type		
Acoustic Suspension System	✓	
Passive Radiator System		✓
Configuration	3-Way	2-Way
Number of Units	3	2
Unit		
Speaker Units		
Woofer (in./cm)	12 (30.5)	8 (20)
Midrange (in./cm)	4-1/2 (11.5)	
Tweeter (in./cm)	1 (2.5) Dome	1 (2.5) Soft dome
Passive Radiator (in./cm)		8 × 12 (20 × 30)
General		
Power Handling Capacity (Music)	150 W (300 W)	130 W (200 W)
Sensitivity (dB/W·m)	91	88
Impedance (ohms)	6	6
Frequency Response (Hz)	40–50,000	35–23,000
Dimensions (W × H × D) in (mm)	15 × 26-3/16 × 13-7/8 (380 × 665 × 351)	13-1/2 × 24-1/2 × 11-5/16 (342 × 622 × 286)
Weight lbs (kg)	62.8 (28.5)	34 (15)

3-way speaker system with 12" cloth-carbon woofer

SX-911WD
3-Way Speaker System

- 12" (30.5 cm) cloth-carbon woofer
- 4-1/2"(11.5 cm) "fine" cloth-carbon midrange
- 1" (2.5 cm) amorphous-diamond coated tweeter
- Low-resonance/vibration die-cast aluminium speaker frames
- Power handling capacity: 150 watts/300 watts (music)

SUPER DIGIFINE

2-way speaker with soft-dome tweeter and passive radiator

SX-A3
2-Way Speaker System

- 8" x 12" (20 x 30 cm) passive radiator
- 2-way design: 8" (20 cm) woofer, 1" (25 mm) soft-dome tweeter
- Self-restoring protection circuit with LED overload indicator
- Magnetic shielding
- Power handling capacity: 130 watts/200 watts (music)

DIGIFINE

Non-directional surround speaker system with black finish

NEW

SP-XS6BK
Surround Speaker System

- Full-range 4-3/4" (12 cm) cone speaker
- High-dispersion design for enlarged listening area
- Floor-standing type
- Power handling capacity: 45 watts/90 watts (music)

SPEAKERS

Non-directional surround speaker system with wood finish

SP-XS5WD
Surround Speaker System

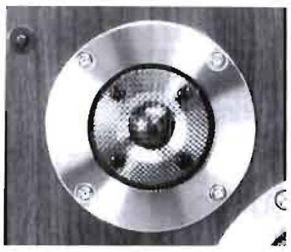
- Full-range 4-3/4" (12 cm) cone speaker
- High-dispersion design for enlarged listening area
- Floor-standing type
- Power handling capacity: 45 watts/90 watts (music)

Technology/Function — **Benefits**

Cloth-carbon diaphragms (SX-911WD)

In our cloth-carbon diaphragm, carbon fiber is woven into the fabric of the diaphragm, to give it greater rigidity. The cloth-carbon diaphragm material specially selected for use in our high-quality woofers has light weight, high rigidity and improved propagation speed; it combines these with optimum internal loss so that low frequency sound is reproduced with greater power.

Since cloth-carbon diaphragms have high rigidity, the oscillations of the voice coil are transmitted more exactly to the diaphragm while the area of the diaphragm which is subject to piston-like motion is greater, without the partial vibrations of less rigid diaphragms. And the cloth-carbon diaphragm material has optimal internal loss, for an improved frequency response throughout the entire range. Moreover, since the upper limit frequency of a cloth-carbon diaphragm is higher than that of a paper cone, a wider range low-frequency sound can be reproduced, so that our cloth-carbon woofer can reproduce powerful, crisp and rich bass sound.



Amorphous-diamond coated tweeter (SX-911WD)

The tweeter of the SX-911WD is coated with a thin layer of amorphous diamond using the high-technology CVD (Chemical Vapor Deposition) process. Featuring uniform thickness, high purity and high surface smoothness, this coating increases the diaphragm's speed of sound propagation to almost that of a natural diamond.

Since the tweeter must reproduce high-frequency sound, it must be made from a light and hard materials. For this reason, we developed amorphous-diamond coating to strengthen the surface. In this construction, a titanium-base diaphragm is coated with a thin layer of amorphous diamond using our original CVD (Chemical Vapor Deposition) technology. The result is an extended frequency response and a dramatically improved transient response.

Soft-dome diaphragm (SX-A3)

Unlike conventional cone-type speakers, soft-dome speakers have a dome-shaped diaphragm made from a soft material (polyester) which has a wider dispersion pattern. Since high frequency sound has a narrow directivity, the area in which it can be heard is limited depending on the size of the speaker cabinet and the listening room. However, JVC's top-end speaker systems employ dome-type diaphragms for their tweeters and midrange speakers, so they are able to be heard over a wider area, due to their wider dispersion pattern.

Although the dome-type diaphragm typically has a lower sensitivity than other types of diaphragm, JVC uses low-mass, pliant polyester for the dome so that the diaphragm moves quickly and with improved sensitivity. The dispersion of the tweeter and the midrange speaker is widened, while the high frequency response is extended and flattened. As a result, crisp and clear high and mid-frequencies are obtained with smooth, natural and coloration-free response. Our soft-dome diaphragms are especially suitable for reproducing sound with a flat and wide frequency response (even including excessive highs), such as from digital sources.

Round-cornered front baffle design (SX-911WD)

The round-corner front baffles of our speaker do more than lend class to overall design. They prevent the diffraction of sound that can occur at sharp edges, causing blurred and indistinct sound images.

Since the round-corner front baffles prevent sound diffraction, you can enjoy clear definition and a lifelike perspective from all our speaker systems.

Self-restoring protection circuit (with LED overload indicator) (SX-A3)

To protect the speaker units from damage due to overloads, a protection circuit is provided just after the input terminals. In the SX-A3, an LED is also provided to indicate that the protection circuit has been activated. Furthermore, the protection circuit is designed to restore itself after the power overload has passed.

This circuit ensures safe operation at excessive power levels. When a possible overload occurs, the signal is interrupted before it is delivered to the speakers. The LED overload indicator warns the user when an overload is imminent, for added convenience. The user can drive speakers even at high power levels without worrying about damage to the speakers.

Magnetic fluid tweeter (SX-A3)

In this model, the voice coil of the tweeter is coated with magnetic lubricant. This gives the tweeter improved heat dissipation characteristics, so that it can handle inputs with higher power and is more resistant to overloads.

Another approach to improved resistance to excessive inputs. Magnetic fluid shielding the tweeter effectively prevents the voice coil from making excessive excursions, while the tweeter diaphragm is protected from external damage. As a result, the tweeter can be driven at higher output levels with no danger of the diaphragm being damaged, while transitions in sound level are handled more smoothly.

Passive radiator (SX-A3)

In the SX-A3, an additional diaphragm is mounted over the port of the bass-reflex cabinet. This speaker diaphragm has no voice coil or magnet and is driven by air pressure (back pressure) in the cabinet, created by the bass driver. A diaphragm of this type is called a "passive radiator".

A passive radiator does not oscillate by itself but is driven by air pressure from the back of the woofer, to oscillate at a frequency around its own resonance frequency (fo) to effectively emphasize low-frequency response. While the "actual" woofer diaphragm radiates solid and rigid bass sound, the passive radiator smooths out and extends the bass response, and produces a greater output than would be obtained with a conventional ported enclosure.

■ Newly developed 3-way Speaker System — SX-911WD

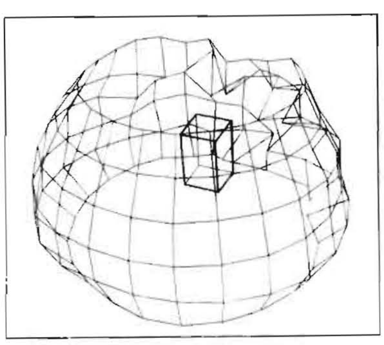
High power handling capacity
150 watts/300 watts (music)

Amorphous-diamond coated tweeter
It features a dome diaphragm with a titanium base on which a thin layer of amorphous diamond is coated using CVD (Chemical Vapor Deposition).

"Fine" Cloth-Carbon midrange speaker
"Fine" Cloth-Carbon in the midrange unit for clear and natural sound.

Cloth-carbon woofer
The newly-developed cloth-carbon woofer is an ideal combination of light weight, high rigidity, high speed of sound propagation and optimum internal loss.

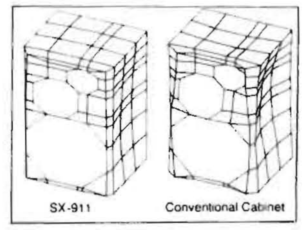
Computer-optimized speaker layout
Thanks to this advanced technique, our speakers combine better definition, smoother frequency response and more accurate phase response.



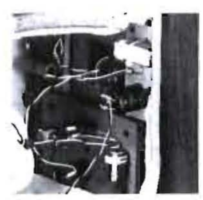
Round-corner front baffleboard
This prevent the sound diffraction that can occur at sharp edges, causing blurred and indistinct sound images.

Propagation Characteristics of SX-911WD
* In addition to the baffle having rounded corners, each part is positioned ideally so a uniform propagation pattern can be obtained in any direction, making the reproduced sound field smooth and rich.

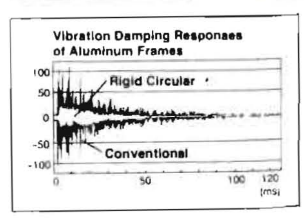
High-density pine-based particleboard enclosure
The panels are made from high-density pine-based particleboard, chosen for its superb musical sonority.



3-part crossover network
In this network system, the high, middle and low frequencies are completely separated to prevent mutual interference so that each of the frequencies is reproduced independently.



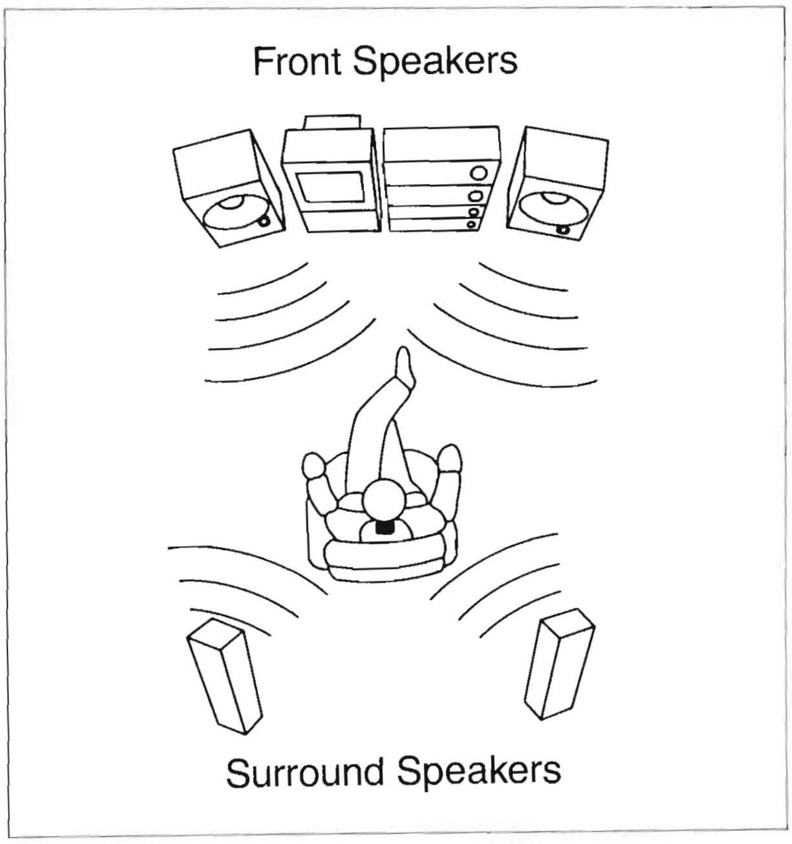
Rigid pure-aluminum frame for woofer
A heavy diecast aluminum frame holds the speaker unit on the front baffle with eight solid screws. This configuration is extremely resistance to resonance.



■ *Non-directional surround speaker systems enhance the feeling of "presence" of the sound field and allow flexible installation*

The SP-XS6BK and SP-XS5WD are surround speaker systems, and both are available in black and wood versions. With compact full-range 4-3/4" non-directional drivers and "tower" cabinets, they are designed for the effective dispersion of sound in even the biggest room. Non-directional characteristics expand surround sound more widely while low-frequency reproduction is excellent thanks to their tower cabinets, giving the ideal "presence" with any sound field. As they have a high 45-watt power handling capacity, these speakers can be used with future amplifiers with high output levels.

As these speakers are compact floor-standing units, the user can locate them easily in the optimum positions of a room, to maximize the surround effect. And because two colors are available, the user can choose the ones that fit in best with his or her interior.



COMPACT COMPONENT SYSTEM

Feature Highlights of the MX-1

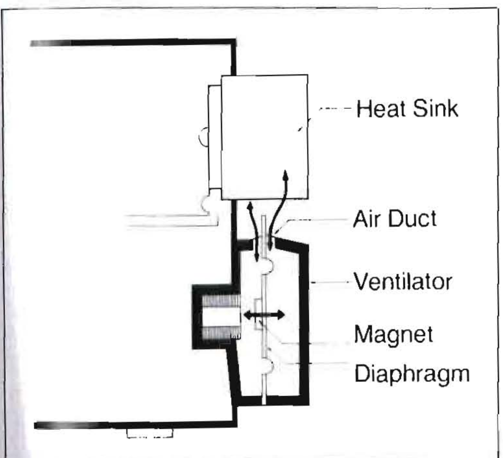
1. *New-concept styling; users can arrange their units to fit in best with any interior design*

2. *Dramatically high power even with compact size, resulting from 3-amplifier configuration and forced cooling system*

3. *Powerful bass from compact speakers with labyrinth aeroport design*

Compact components designed for maximum flexibility in installation

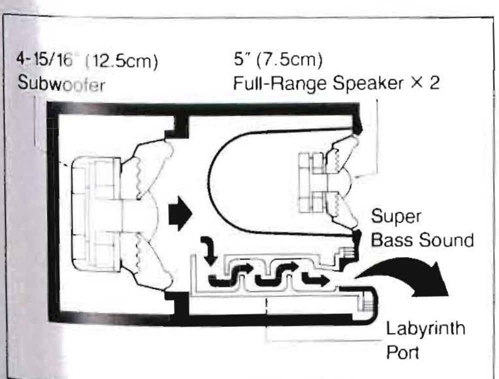
These compact components are styled to be interior design accessories, wherever you install them, in your living room, bedroom or den, and however you arrange them, horizontally or vertically, they'll complement the room's interior. Create your own style, use these compact components as part of your own modern life style.



You'd never believe such powerful sound could come from such compact components

In the 3-amp configuration, in addition to powerful amplifiers driving the left and right speakers, there's another amplifier provided to drive the subwoofers. An amplifier with a high power output generates a great deal of heat. To dissipate this heat we developed a new forced cooling system, with a ventilator cooling the heat sink. As a result, this arrangement allows the size of the heat sink to be reduced to one quarter that of a conventional heat sink providing the same heat dissipation effect, which in turn allows a pair of compact aeroport speakers to deliver the high total power of 100 watts*.

* Main amplifier: 2 × 30 watts at no more than 0.9% THD (8 ohms, 20 Hz – 20 kHz) (RMS)
Subwoofer amplifier: 40 watts at no more than 0.9% THD (6 ohms, at 80 Hz)



Powerful bass sound from such compact speaker systems

To reproduce bass more powerfully, the size of woofers must be increased, or the length of the ducts extended. However, in a compact component system, increasing the size of the woofers is impossible. To overcome this problem, JVC provides a 12.5-cm subwoofer inside the speaker cabinet which is driven by a third amplifier exclusively for very low frequencies and a new labyrinth duct which has sufficient effective length to enhance the output of the subwoofer. With this system, high-quality sound with dramatically improved reproduction at extremely low frequencies is possible from speakers with compact size.

Feature Comparison Chart

Compact Component System

		MX-1
General		
Remote Control Unit		✓ (RM-SEMX1)
CD Player and Amplifier Section		
Power Output*		2 × 30 W (Main), 40 W (Subwoofer)
Motor-Driven Volume		✓
3-Amp Configuration with forced cooling system		✓
SEA	**Electronic**	✓
	No. of Elements	7
Single Tray		✓
3-Beam Laser Pickup		✓
Digital Filter		2fs, 16-bit
3-Way Edit with Fade		✓
Numeric Keys (Remote)		✓
Program Chart Display (Calendar)		✓ (20)
No. of Programs		32
Timer Play		✓
Search (Auto/Manual)		✓
Repeat Play		✓
Hi-Balance (New Y-Servo)		✓
ISS		✓
Tuner and Cassette Deck Section		
Preset Stations (Random)		40
Preset Scan		✓
Auto Memory		✓
5-Way Program Timer		✓
FL Display		✓
Double-Mechanism		✓
Deck A (Play)		✓
Deck B (Record/Play)		✓
Full-Logic Control		✓
Auto-Reverse		✓
High-Speed Dubbing (Deck A-B)		✓
Continuous Play		✓
Music Scan		✓
Electronic Counter		✓
Rec Mute		✓
Dolby NR		B
Speaker Section		
Labyrinth Aeroport System		✓
Speaker Units	Full-Range	7.5cm × 2 (Cone)
	Subwoofer	12.5cm (Cone)
Power Handling Capacity, Rated/Music		45 W (Main)/30 W (Subwoofer)

* Main: 2 × 30 W at no more than 0.9% THD (8 ohms, 20 Hz – 20 kHz) (RMS)
Subwoofer: 40 W at no more than 0.9% THD (6 ohms, at 80 Hz)

Remote Control Unit

		RM-SEMX1
General		
Provided with		MX-1
Power ON/OFF		✓
Volume (Up/Down)		✓
Sleep/Wake-up		✓
Fade Mute		✓
Daily Timer		✓
Source		
Tuner		✓
CD		✓
Tape		✓
VCR		✓
DAT		✓
AUX		✓
Tuner		
Direct Access (Numeric Keys)		✓
CD Player		
Numeric Keypad (1-10, +10, 0)		✓
Play/Stop		✓
Auto Search		✓
Manual Search		✓
Open/Close		✓
Auto/Program Edit		✓
Cassette Deck		
Deck A	Play (FWD/REV)	✓
	Stop	✓
	FF/Rew	✓
Deck B	Play (FWD/REV)	✓
	Stop	✓
	Rec	✓
	Pause	✓
	FF/Rew	✓
SEA		
SEA ON/OFF		✓
Frequency Adjust Control		✓
SEA Preset		✓
DAT		
Play		✓
Stop		✓
Rec		✓
Pause		✓
FF/REW		✓
VCR		
Play		✓
Stop		✓
Rec		✓
Pause/Still		✓
FF/REW		✓
10 Key Keypad		✓

Remote-controlled compact component system with the extremely high total power of 100 W and labyrinth aeroport speakers*

MX-1
Compact Component System

- **Power output: 2 x 30 watts + 40 watts***
- **3-amp high-power design**
- **Newly-developed forced air cooling system**
- **Labyrinth aeroport speakers**
- **3-way editing function; auto, programmed and fade**
- **Computer-controlled S.E.A. graphic equalizer**
- **5-way program timer with 5-step volume**

*Main 2 x 30 W at no more than 0.9% THD (8 ohms, 20 Hz - 20 kHz) (RMS)
Subwoofer 40 W at no more than 0.9% THD (6 ohms, at 80 Hz)

■ *New-concept styling to match any life style*

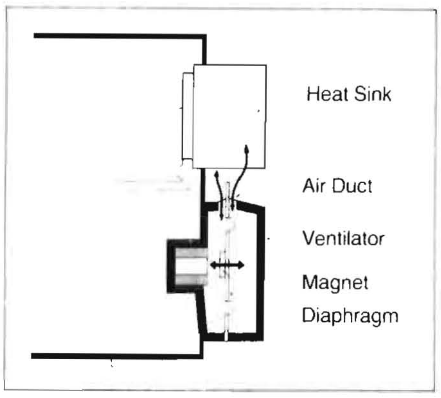
This compact component system has been developed from a completely new concept. Even though the components are compact, their high power has not been compromised. These fashionable components are ideal for installation where space is at a premium, in bedrooms, dens, on a shelf, under a bed. As these components are designed especially for both horizontal or vertical placement and stacking, you can arrange them as required to complement your room's interior, without worrying about how much space they'll take up.

3-amplifier configuration

In addition to a two-channel stereo amplifier, a third amplifier is provided to drive the subwoofers, to make the most of digital programs with their wider dynamic range. With this system, a pair of compact aeroport speakers can deliver the high total power of 100 watts*.

* Main: 2 x 30 W at no more than 0.9% THD (8 ohms, 20 Hz — 20 kHz) (RMS)
Subwoofer: 40 W at no more than 0.9% THD (6 ohms, at 80 Hz)

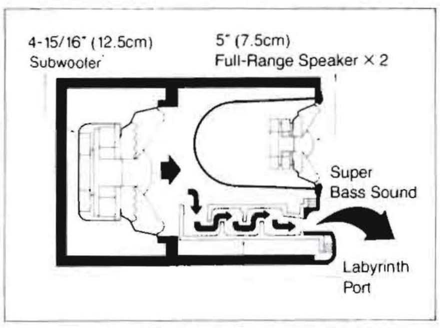
As a third amplifier is provided exclusively to drive the subwoofers, this compact system is able to reproduce low frequencies with remarkable power.



Forced cooling system

This system uses a newly-developed forced cooling system with a ventilator below the heat sink providing a cooling flow of air which dissipates any build up of heat; this allows the size of the heat sink to be reduced to only one quarter the size of that in a conventional amplifier with comparable power.

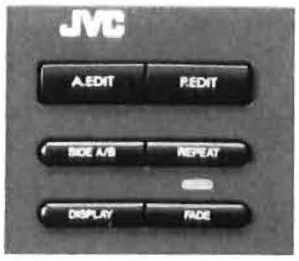
The cooling system only operates when a sensor detects that the temperature of the heat sink has risen above a certain level. When activated, the ventilator's diaphragm oscillates at a very low frequency; this frequency is too low to produce any interference with the music, thereby a possible source of noise is eliminated, while the cooling effect is optimized. Thanks to the new forced cooling system reducing the size of the heat sink and allowing a compact amplifier to deliver more powerful sound, the overall size of the system is smaller, for more flexible installation.



Labyrinth port system

To reproduce bass more powerfully, the 12.5-cm subwoofers in each of the two speaker cabinets, driven by a third amplifier, use new labyrinth ducts with sufficient effective length to enhance the output low-frequency sound.

Generally, the size of woofers must be increased or the length of their ducts extended for the more powerful reproduction of bass frequencies. To achieve this while minimizing size, JVC uses a labyrinth port system; the labyrinth ducts resonate with the sound generated by the subwoofers, effectively extending the length of the ducts. So, despite their compact size, the speakers can deliver rich bass sound with the ideal damping.



Auto/program/fade editing function

The auto editing function allows you to record tunes from a compact disc in the order they're recorded on the disc according to the tape length which is previously set by the user; if there is insufficient time left on the tape for the next tune, the microcomputer searches for tunes that will fit into the time remaining and shows their tune numbers so that the user can specify the tune to be recorded, for more efficient use of the tape. With the program editing function, you can record tunes in any required order and assign which side of the tape (A or B) they should be recorded on. With the fade editing function, when the end of side A of a tape is reached while a tune is still being recorded, the tune fades out and the whole of that tune is re-recorded at the start of side B. For the most effective use of these three editing functions, the fade function can be used together with the auto or program editing function.

With the auto and program functions, you can create original, customized tapes; the fade editing function eliminates the unpleasant phenomenon of music being interrupted abruptly. You can also use fade editing to maximize use of your tapes by recording part of a tune and fading it out just before the end of the tape is reached.

Refer to "Demonstration" page 103 for more information.

5-way programmable timer with 5-step volume setting

The programmable timer operates in 5 modes, WAKE-UP, SLEEP, TIMER-1, TIMER-2 and DAILY. In the TIMER-1, TIMER-2 and DAILY modes, the volume can be set by selecting one of four factory-preset levels and one user-preset level, for each source; at the preset time, the required program is played back with its level gradually increased to the level selected.

There are so many ways you can use this convenient facility to enhance your life. In the TIMER-1 and TIMER-2 modes, you can preset the times the playback or recording of any source is to start and end. In the DAILY mode, you can listen to or record the same source at the same time every day without having to preset the timer each day. With the WAKE-UP mode, you can be woken up with the selected source at the desired volume In the SLEEP mode, as the power is switched off automatically after a specified time, you don't have to worry about forgetting to switch the power off when you listen in bed just before you fall asleep. In addition, as the volume can be set independently for each source, you don't have to worry about the different output levels of different sources and you can set whatever volume you want for each. Further, as the volume is increased gradually in the TIMER-1, TIMER-2 and DAILY modes, the source selected to be played back need not wake you with a sudden blast of sound, it can wake you up in the gentlest way.

Refer to "Demonstration" page 102 for more information.

Six programmed and one user preset equalization pattern

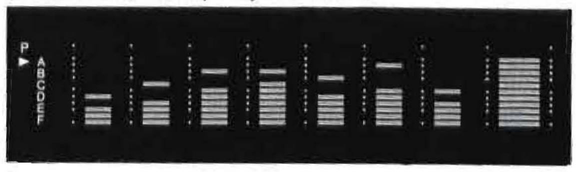
Six programmed equalization patterns — ROCK, JAZZ, POPS, CLASSIC, HEADPHONE and CAR — can be recalled instantly while you can create one equalization pattern yourself and store it in memory. According to the source you select, you can call up any of these patterns by simply pressing a button.

The seven preset equalization patterns can be recalled by just touching a button; they can then be adjusted with the buttons on the main unit or the remote control unit to obtain any required effect.

Refer to "Demonstration" page 102 for more information.

7-band computer-controlled S.E.A. graphic equalizer with 6 programmed patterns and 1 user-programmable pattern

The MX-1 has 6 programmed patterns and 1 user-programmable pattern and these patterns can be recalled and fine adjusted not only with the front panel controls, but also using the remote control unit. Use a variety of patterns according to the situation and the customer's taste in music, to perform demonstrations. The SPI — spectro peak indicator — display shows seven constantly-changing bars each representing the level of the signal in one of seven frequency bands with the peak level of each frequency band held for several seconds; this eye-catching display makes the demonstration even more attractive.



■ 6 programmed equalization patterns

Briefly, these programmed equalization patterns are as follows.

ROCK: Adds power to rock, with heavy bass and crisp highs.

JAZZ: Ideal for the reproduction of live recordings, making the most of the tonal qualities of acoustic musical instruments.

POPS: Makes pop music more attractive by emphasizing frequencies in the vocal register

CLASSIC: Orchestral music is enhanced by "expanding" the sound.

H.PHONE: This pattern gives sound heard with headphones greater clarity.

CAR: Record cassettes to be played back by a car audio system for better sound in a restricted space.

■ Recalling the preset equalization patterns

The programmed and user-programmable preset patterns can be recalled by simply touching a button. To show the preset patterns, follow the procedure below.

1) Press the SEA button so that the SEA indicator lights.
2) Press the SEA PRESET button; each time this is pressed, the preset patterns are shown in sequence.
3) So that customers can hear how much difference the S.E.A. graphic equalizer makes, switch it off by pressing the SEA button (the indicator goes out), then switch it on again.

Versatile 5-Way Program Timer with 5-Step Volume

The program timer operates in 5 modes WAKE-UP, SLEEP, TIMER-1, TIMER-2 and DAILY. With the WAKE-UP mode, you can be woken up at the preset time by your preferred source, with the volume set to the desired levels.

With the SLEEP mode, the power is switched off automatically after a specified time, even if you've fallen asleep.

And with the TIMER-1 and TIMER-2 modes, you can program the times the source selected is to start and end playback or recording.

The DAILY mode lets you to listen to or record the same source at the same time every day, without you're having to preset the timer each day.

Further, in the TIMER-1, TIMER-2 and DAILY modes, any of four volumes can be selected — these are shown in the display as VOL 0 (when you don't want to monitor the recording), VOL A, VOL B, VOL C — plus an additional volume you can set yourself. When the source you've selected starts playing, the volume gradually increases to this preset level. This prevents the music starting with a sudden blast of sound and wakes the user up more gently than the usual alarm.

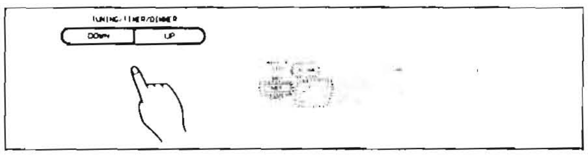
For an effective demonstration of these functions, we recommend that you use the TIMER-1 function and choose a CD as the source; this allows you to demonstrate the setting of the timer and the gradually increasing volume as an especially attractive feature.

(TIMER-1 function; to play back a CD)

1) Switch the amplifier section ON.

2) Press (TIMER 1).

3) Press (TUNING/TIMER/DIMMER) to set the hours of the start time. (The time is shown using a 12-hour clock, with AM/PM indicators.) Then, press (MEMORY).



4) Press (TUNING/TIMER/DIMMER) to set the minutes of the start time and then press (MEMORY).

5) Press (TUNING/TIMER/DIMMER) to set the hours of the stop time and then press (MEMORY).

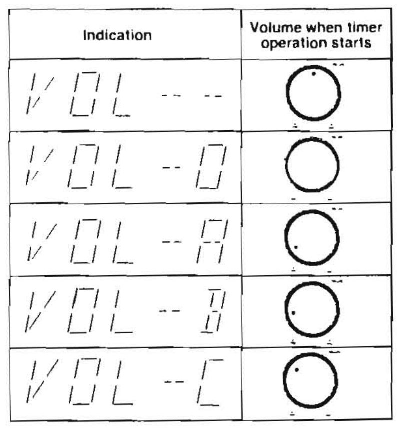
6) Press (TUNING/TIMER/DIMMER) to set the minutes of the stop time and then press (MEMORY).

7) Press (TUNING/TIMER/DIMMER) so that the source to be played back is shown in the display. In this case, press until "CD" is shown. Now press (MEMORY).

8) Press (TUNING/TIMER/DIMMER) to show the volume in the display; you can set the most suitable volume.

9) Press (TIMER 1) and press POWER again to switch off the power.

After this, when the preset time is reached, the CD will be played back with the volume gradually increasing at the start.

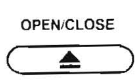


Convenient auto/program/fade editing function

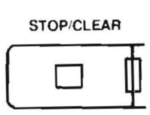
The MX-1 is provided with auto editing, programmed editing and fade editing functions. Combining the programmed editing and fade editing functions is an effective demonstration, showing how easy it is to make a customized tape.

1) Load a cassette in deck B.

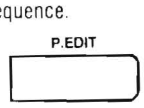
2) Load a CD after pressing (OPEN/CLOSE).



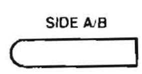
3) Press (STOP/CLEAR) of the CD section.



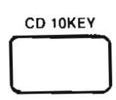
4) Press (P.EDIT) to set the type of tape; C46, C54, C60, C74 and C90 are shown in sequence.



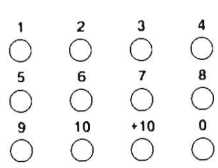
5) Specify the side to be recorded by pressing (SIDE A/B).



6) Press (CD 10KEY).

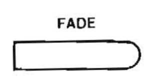


7) Use the numeric keys to specify the order in which tunes are to be recorded.

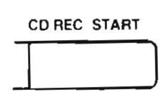


8) To record on the other side of the tape, repeat steps 5 — 7

9) Press (FADE) of the CD player section.



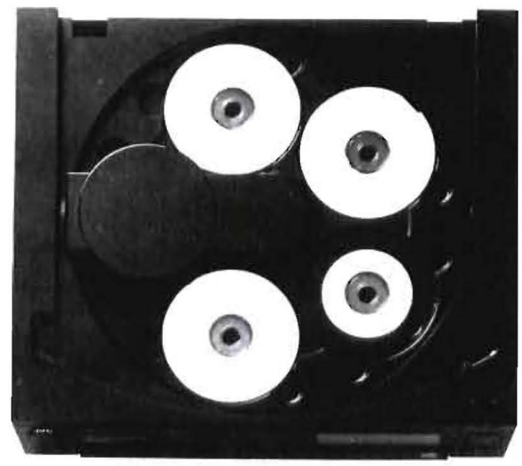
10) Press (CD REC START) of the cassette deck section.



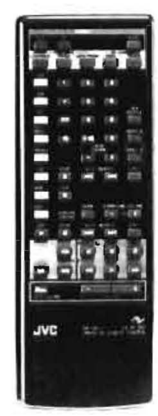
With these procedures, the selected tunes are recorded onto the cassette in the specified order, with the last tune fading out.

Feature Highlights of '90 JVC Stylistic Hi-Fi Systems

1 *Carousel CD auto-changers provided in two systems*



2 *Remote control units are provided as standard with all systems*



3 *Most system have an attractive black finish*

Feature Comparison Chart
JVC Stylistic Hi-Fi Systems

Feature		GX-8800CDM	GX-8700CDM	GX-8500CDM	GX-8400CDM	GX-930CD
AV Remote		✓	✓	✓	✓	✓
Amplifier		AX-R87BK	AX-R87BK	AX-R87BK		
Power Output (watts)		125 W	125 W	125 W		
THD (%), Rated Output Power		0.03	0.03	0.03		
COMPU LINK Remote Control		✓	✓	✓		
Motor-Driven Volume		✓	✓	✓		
SEA Graphic Equalizer	Mechanical	✓	✓	✓		
	No. of Elements	7	7	7		
Gm Driver		✓	✓	✓		
LED Peak Indicator		✓	✓	✓		
Source	CD/Tuner/Phono/Tape 1/Tape 2/VCR/TV	✓	✓	✓		
Tuner		FX-87BK	FX-87BK	FX-87BK		
COMPU LINK Component		✓	✓	✓		
Preset Stations (Random)		40	40	40		
Auto Memory		✓	✓	✓		
Preset Scan		✓	✓	✓		
Receiver					RX-R85BK	RX-R85BK
Power Output (watts)					100 W	100 W
THD (%), Rated Output Power					0.03	0.03
COMPU LINK Remote Control					✓	✓
Motor-Driven Volume					✓	✓
SEA Graphic Equalizer	Mechanical				✓	✓
	No. of Elements				5	5
Gm Driver					✓	✓
Source	CD/Tuner/Phono/Tape 1/Tape 2(Video Sound)				✓	✓
Preset Stations (Random)					40	40
Auto Memory					✓	✓
Preset Scan					✓	✓
CD Player		XL-M87BK	XL-M87BK	XL-R86BK	XL-R86BK	XL-V85BK
Type	Auto Changer	✓	✓			
	Carousel			✓	✓	
	Single Tray					✓
COMPU LINK Component		✓	✓	✓	✓	✓
Digital Filter		✓ (4fs)	✓ (4fs)	✓ (4fs)	✓ (4fs)	✓ (2fs)
ISS		✓	✓	✓	✓	✓
No. of Programs		32	32	32	32	32
Repeat		✓ (4-way)	✓ (4-way)	✓ (2-way)	✓ (2-way)	✓ (3-way)
FL Display		✓	✓	✓	✓	✓
Cassette Deck		TD-W87BK	TD-W87BK	TD-W85BK	TD-W83BK	TD-W83BK
COMPU LINK Component		✓	✓	✓		
Full-Logic Control		✓	✓	✓		
Deck A (Play)		✓	✓	✓	✓	✓
Deck B (Rec/Play)		✓	✓	✓	✓	✓
Auto-Reverse		✓	✓			
High-Speed Dubbing (Deck A-B)		✓	✓	✓	✓	✓
Synchro Dubbing (Deck A-B)		✓	✓	✓	✓	✓
Dolby NR		B/C	B/C	B	B	B
Turntable		AL-F97BK	AL-F97BK	AL-A95BK		AL-A95BK
COMPU LINK Component		✓	✓			
Full Automatic		✓	✓			
Semi-Automatic				✓		✓
Belt Drive		✓	✓	✓		✓
Auto Size Selector		✓	✓			
Auto Speed Selector		✓	✓			
Plug-In Cartridge (T4P)		✓	✓			
Speaker		SP-87BK	SP-87BK	SP-85BK	SP-83BK	SP-93WD
Configuration		3-Way	3-Way	3-Way	3-Way	3-Way
Bass-Reflex				✓	✓	✓
Woofer (inch)		12	12	12	10	10
Midrange (inch)		4	4	4	4	4
Tweeter (inch)		3-1/8	3-1/8	3-1/8	3-1/8	3-1/8
Passive Radiator (inch)		✓ (12)	✓ (12)			
Power Handling Capacity, Rated/Music (watts)		125/210	125/210	125/210	100/180	100/180
Hybrid Olefin Cone		✓	✓			
Low-leakage Magnet Design		✓	✓	✓	✓	✓
Round Baffle Design		✓	✓	✓		
4-Layer Voice Coil		✓	✓	✓	✓	✓
Color Finish		Black	Black	Black	Black	Wood
Rack		RK-87BK	RK-87BK	RK-86BK	RK-86BK	RK-95WD
Tower Type		✓	✓	✓	✓	✓
2-Glass Doors		✓	✓			
Color Finish		Black	Black	Black	Black	Wood
Surround		SU-A97				
Surround Amp	Power Output (watts)	10				
	Dolby Surround	✓				
		SP-XS6BK				
Surround Speaker	Full-Range (inch)	4-3/4				
	Power Handling Capacity, Rated/Music (watts)	45/90				

Feature Comparison Chart

Remote Control Unit

		RM-SA87U	RM-SR85U
General			
Provided with		AX-R87BK	RX-R85BK
Power ON/OFF		✓	✓
Volume (Up/Down)		✓	✓
Fade Muting		✓	✓
Source			
FM		✓	✓
AM		✓	✓
CD		✓	✓
TAPE 1		✓	✓
TAPE 2		✓	✓
Phono		✓	
VCR		✓	✓
TV		✓	
SEA			
SEA Source		✓	
Tuner			
Direct Access (Numeric Keys)		✓	✓
Preset Stations Up/Down		✓	
CD Changer			
Disc Select		✓	✓
Track Select (10-key)		✓	✓
Play Mode	Continue	✓	✓
	Program	✓	✓
CD Player			
Numeric Keypad (1-10, +10, 0)		✓	✓
Play/Stop		✓	✓
Auto Search		✓	✓
Surround			
Surround Volume		✓	
Mode		✓	
Turntable			
Play		✓	
Stop		✓	
Cassette Deck (Tape)			
Play		✓	
Stop		✓	
Rec		✓	
Pause		✓	
FF/REW		✓	
DAT			
Play (10 Key Keypad)		✓	
Stop		✓	
Rec		✓	
Pause		✓	
FF/REW		✓	
VCR			
Play		✓	✓
Stop		✓	✓
Rec		✓	✓
Pause/Still		✓	✓
FF/REW		✓	✓
10 Key Keypad		✓	
Channel Up/Down		✓	
TV			
10 Key Keypad		✓	
Channel Up/Down		✓	
Video 2		✓	
Video 3		✓	

Top-of-the-line COMPU LINK A/V remote controlled hi-fi system featuring A/V Surround system

NEW

Black-finished

GX-8800CDM

COMPU LINK Remote Control System

Stylistic Hi-Fi System

AX-R87BK
Remote-controlled integrated amplifier
FX-87BK
Computer-controlled digital synthesizer tuner
XL-M87BK
Compact disc auto changer
TD-W87BK
Full-logic control double-mechanism cassette deck with Hi-Fi U-Turn auto-reverse
AL-F97BK
Fully-automatic turntable
SP-87BK
3-way speaker system with passive radiator
SU-A97
A/V surround processor
SP-XS6BK
Surround speaker system
RK-87BK
Audio rack

High-power COMPU LINK remote control hi-fi system with CD auto changer and bass-extended 3-way speaker system

NEW

Black-finished

GX-8700CDM

COMPU LINK Remote Control System

Stylistic Hi-Fi System

AX-R87BK
Remote-controlled integrated amplifier
FX-87BK
Computer-controlled digital synthesizer tuner
XL-M87BK
Compact disc auto changer
TD-W87BK
Full-logic control double-mechanism cassette deck with Hi-Fi U-Turn auto-reverse
AL-F97BK
Fully-automatic turntable
SP-87BK
3-way speaker system with passive radiator
RK-87BK
Audio rack

High-power COMPU LINK remote control hi-fi system with carousel CD auto-changer

NEW

Black-finished

GX-8500CDM

COMPU LINK Remote Control System

Stylistic Hi-Fi System

AX-R87BK
Remote-controlled integrated amplifier
FX-87BK
Computer-controlled digital synthesizer tuner
XL-R86BK
Carousel CD auto-changer
TD-W85BK
Full-logic control double-mechanism cassette deck
AL-A95BK
Auto-return turntable
SP-85BK
3-way speaker system
RK-86BK
Audio rack

High-power COMPU LINK remote control hi-fi system with carousel CD auto-changer

NEW

Black-finished

GX-8400CDM COMPU LINK Remote Control System

Stylistic Hi-Fi System

RX-R85BK
Computer/remote-controlled receiver

XL-R86BK
Carousel CD auto-changer

TD-W83BK
Double-mechanism cassette deck

SP-83BK
3-way speaker system

RK-86BK
Audio rack

High power COMPU LINK remote control hi-fi system with CD player

NEW

Wood-finished

GX-930CD COMPU LINK Remote Control System

Stylistic Hi-Fi System

RX-R85BK
Computer/remote-controlled receiver

XL-V85BK
Compact disc player

TD-W83BK
Double-mechanism cassette deck

AL-A95BK
Auto-return turntable

SP-93WD
3-way speaker system

RK-95WD
Audio rack

NEW **Amplifier**

AX-R87BK

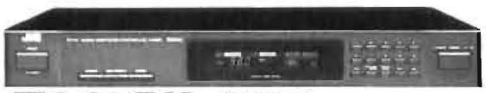
COMPU LINK Remote Control Component

Remote Control Integrated Amplifier

- Power output: 2 x 125 watts*
- Unified A/V remote control
- 7-band S.E.A. graphic equalizer
- 7-segment LED power output indicator
- Motor-driven volume control with indicator

*at no more than 0.03% THD (8 ohms, 20 Hz — 20 kHz) (RMS)

NEW **Tuner**



FX-87BK

COMPU LINK Component

Computer-Controlled Digital Synthesizer Tuner

- Preset memory of 40 FM/AM stations
- Preset scan
- Auto memory

NEW **Receiver**

RX-R85BK

COMPU LINK Remote Control Component

Remote Control Receiver

- Power output: 2 x 100 watts*
- 5-band S.E.A. graphic equalizer
- Preset memory for 40 FM/AM stations
- Full-function remote control
- 7-segment LED power level indicator

*at no more than 0.03% THD (8 ohms, 20 Hz — 20 kHz) (RMS)

NEW **Compact disc players**

XL-M87BK

DIGIFINE COMPU LINK Component

Compact Disc Auto Changer

- CD auto changer with 6-disc magazine
- 4-times oversampling digital filter
- Dual D/A converter
- Random access programming of up to 32 steps
- Random play

NEW

XL-R86BK

DIGIFINE COMPU LINK Component

Carousel 5-Disc Compact Disc Changer

- 4-times oversampling digital filter
- Dual D/A converters
- Random play, continuous play and program play
- Multi-function display
- Random access programming of up to 32 steps

NEW

XL-V85BK

DIGIFINE COMPU LINK Component

Compact Disc Player

- 2-times oversampling digital filter
- Dual D/A converters
- Random access programming of up to 32 tracks
- 2-way editing function
- 3-way repeat

Surround processor

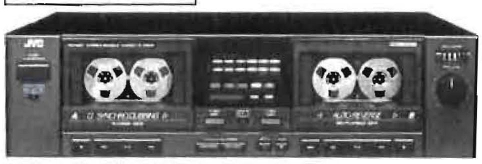
SU-A97
A/V Surround Processor

- Dolby Surround, Hall Surround and Simulated Surround
- Built-in rear-channel power amplifier (10W + 10W)*

*at no more than 0.5% THD (8 ohms from 40 Hz — 20 kHz) (RMS)

NEW

Cassette decks



TD-W87BK
COMPU LINK Component

Full-Logic Control Double-Mechanism Cassette Deck with Hi-Fi U-Turn Auto-Reverse

- Hi-Fi U-Turn auto-reverse with Flip Reverse Head (Deck B)
- Record/play and play-only tape transports
- Computer-controlled full-logic control
- Continuous play of two tapes
- Dolby B/C noise reduction

NEW

TD-W85BK
COMPU LINK Component

Full-Logic Control Double-Mechanism Cassette Deck

- Record/play and play-only tape transports
- Computer-controlled full-logic control
- Continuous play of two tapes
- High-speed editing with synchro start
- Dolby B noise reduction

NEW

TD-W83BK
Double-Mechanism Cassette Deck

- Record/play and play-only tape transports
- Continuous play of two tapes
- High-speed editing with synchro start
- Dolby B noise reduction

Turntables

AL-F97BK
COMPU LINK Component

Fully Automatic Belt-Drive Turntable

- Fully automatic operation
- Auto record size and speed selector
- Plug-in moving-magnet cartridge

AL-A95BK
Auto-Return Belt-Drive Turntable

- Low-mass straight tonearm
- Automatic arm return

Speaker systems

SP-87BK

3-Way Speaker System with Passive Radiator

- 3-way tower design with 12-inch woofer
- 12-inch passive radiator
- 4-layer woofer voice coils with magnetic fluid in gaps
- Round-cornered front baffle

SP-85BK

3-Way Speaker System

- 3-way tower design with 12-inch woofer
- 4-layer woofer voice coils with magnetic fluid in gaps
- Round-cornered front baffle

SP-83BK

3-Way Speaker System

- 3-way tower design with 10-inch woofer
- 4-layer woofer voice coils with magnetic fluid in gaps
- Heat-resistant adhesive

SP-93WD

3-Way Speaker System

- 3-way tower design with 10-inch woofer
- 4-layer woofer voice coils with magnetic fluid in gaps
- Heat-resistant adhesive

Surround speaker system

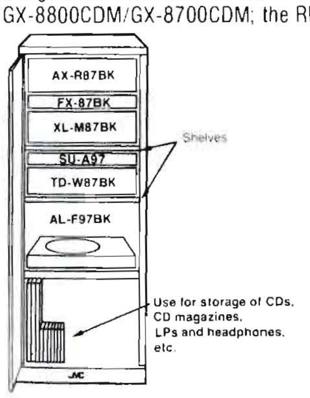
SP-XS6BK

Surround Speaker System

- Full-range 5-inch cone speaker
- High-dispersion design for enlarged listening area

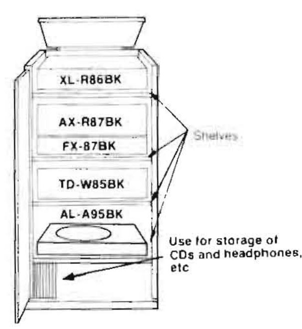
Examples showing how to install in the rack

The RK-87BK audio rack is specially designed for installation of the JVC Stylistic GX-8800CDM/GX-8700CDM; the RK-86BK audio rack is for the Stylistic GX-8500CDM/GX-8400CDM; the RK-95WD audio rack is for the Stylistic GX-930CD. The following shows the examples of the most effective ways for each system.



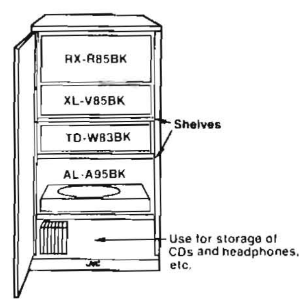
For installing Stylistic GX-8800CDM in the RK-87BK

1. Install the shelves as shown. The turntable is installed on the bottom shelf, the cassette deck and the A/V surround processor on the next, and the CD changer, tuner and amplifier are stacked on the top shelf.
2. Install the components as shown in the figure.



For installing Stylistic GX-8500CDM in the RK-86BK

1. Install the shelves and components as shown in the figure.



For installing Stylistic GX-930CD in the RK-95WD

1. Install the shelves as shown. The turntable is installed on the bottom shelf, the cassette deck on the next, and the CD player and receiver are stacked on the top shelf.

MEMO

New Hi-Fi Technology 1990

JVC

JVC-developed PEM DD (Pulse Edge Modulation Differential-Linearity-Errorless DA) converter

1) Are DA converters using more bits required?

Basically, the DA converter determines the quality of sound from a digital source, and the main competition to develop better DA converters has been in the number of bits used in conversion, from 16 bits, to 18 bits, 20 bits and even 22 bits.

For example, the signals recorded on a compact disc are 16-bit, so why is the number of bits increased above this? The main reason is to reduce degradation of the S/N introduced by the digital filter before the DA converter. This digital filter is used to reduce the complexity of the analog filter after the DA converter, thereby improving the phase characteristics. However, the digital filter degrades the S/N because requantization noise (rounding-off errors) is added to the quantization noise which is inherent in the system. In spite of this drawback, the digital filter is used to increase the sampling frequency and simplify the analog filter, for higher quality sound.

There are two ways in which the degradation of the S/N can be reduced; by increasing the multiplying factor of oversampling and by increasing the number of bits. While the S/N is improved as the multiplying factor of oversampling and the number of bits are increased, there are limits. With 4 fs oversampling and 18 bits, the S/N becomes close to the ideal value, that is, quantization noise generated in the recording process cannot be removed, however much the reproduction system is improved. So, increasing the values above these figures has no effect on the S/N. Therefore, whichever of these two methods, or whatever combination of them, is adopted, there are limitations. Furthermore, even when the number of bits is increased, only the S/N of the signal output from the digital filter can approach the ideal value. With regard to the entire DA converter, the S/N is not improved unless the accuracy of the DA converter is equivalent to the number of bits.

2) Operation of a multi-bit DA converter (Fig. 1)

The digital signals recorded on a compact disc (as PCM signals) are a train of data representing the amplitude of the signal in a fixed time interval (the sampling period). If the bits making up this amplitude are replaced by voltages or currents proportional to the values of the individual bits, using the same sampling frequency as that used for AD conversion, the digital signals can be converted into analog signals. This principle is used in conventional multi-bit DA converters; as the currents are produced by a "resistance ladder", these DA converters are generally called "ladder-type" DA converters. In the most popular ladder type DA converters, there are a number of constant current sources corresponding to each bit, from the LSB (least significant bit) to the MSB (most significant bit), the weighting of each of which is double that of the next lower bit. According to the instantaneous digital data, each current source is

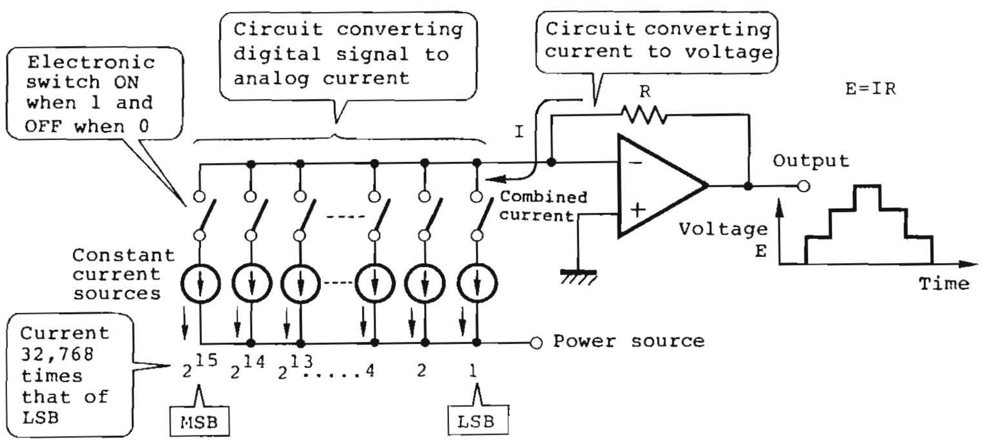


Fig. 1 Principle of 16-bit current-summing type DA converter

switched ON or OFF to generate a current the total amplitude of which corresponds to the instantaneous value of the data, then this current is converted into a voltage.

As the signals produced by the above process are step-shaped, an analog LPF (Low Pass Filter) is used to remove pulse components and output smooth analog signals.

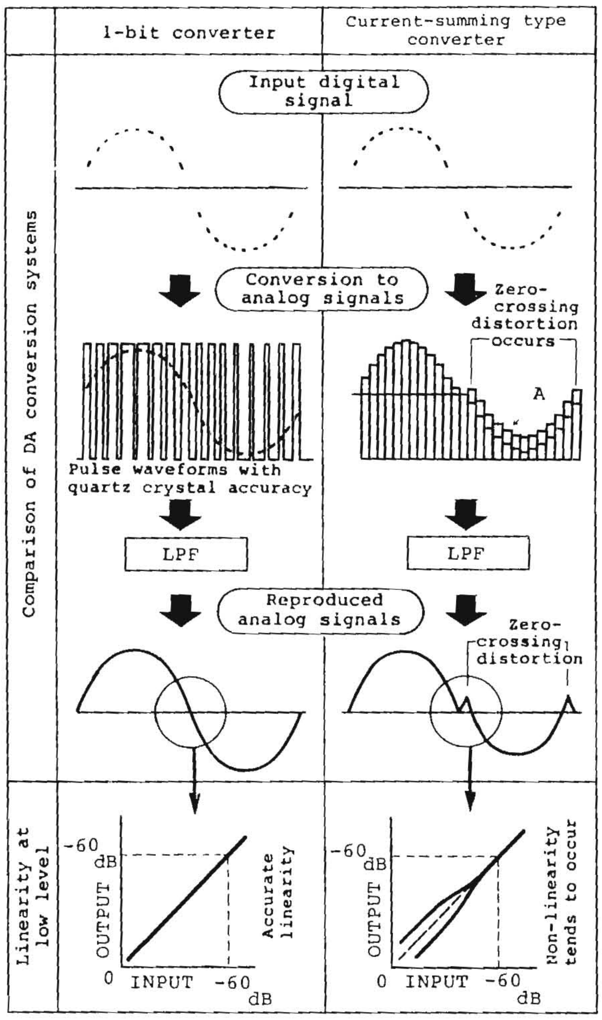
A current-summing type DA converter uses a ladder of at least 16 resistance elements and the value of the largest of these is 32,768 times that of the smallest, but it is difficult to accurately provide resistances with such a large difference; the problem is that any error in these resistances results in a differential linearity error. These resistances are influenced by changes in temperature and aging, which again makes it difficult to maintain accuracy. Especially, this error is noticeable around the 0 point of the analog music signal, that is, the point where only the MSB is on. When the signal drops 1 step below this value, the MSB switches off and all the other bits switch on; if the values of the resistance elements are not aligned with extreme precision, this produces a phenomenon called "zero-crossing" distortion because it occurs at the point where the analog signal crosses the time axis.

And as these 16 resistances are switched independently, if the timing of the switching is less than perfect, it can introduce another form of noise, "glitches", which result in waveform distortion.

3) Operation of a 1-bit DA converter

On the other hand, the 1-bit DA converter has only 2 states, 1 and 0, high and low, or ON and OFF. Switching between these two states is performed at an extremely high speed and analog signals are formed by varying the time for which the circuit is ON and OFF, which corresponds to the digital data from the source.

In a 1-bit DA converter, there is only one amplitude, and it does not rely on the accurate alignment of the weighted values of a ladder of resistance elements which are vulnerable to changes in temperature and aging; as the switching timing is controlled by the high-precision clock generated by a quartz crystal oscillator, low distortion can be obtained without any adjustment while zero-crossing distortion and glitches do not occur due to the circuit's principles of operation. This is what gives a 1-bit DA converter the ability to reproduce very low level signals accurately, so as to reproduce the "presence" of a sound field and musical nuances with extreme fidelity. These are the most important advantages of a 1-bit DA converter (Fig. 2).



A. Amplitude changed due to bit errors

Fig. 2 Comparison of 1-bit DA converter and current-summing type DA converter

4) JVC PEM DD converter (Fig. 3)

The PEM DD converter is an original JVC invention and consists of a newly-developed fourth-order noise shaper and a "1-bit PEM DA converter".

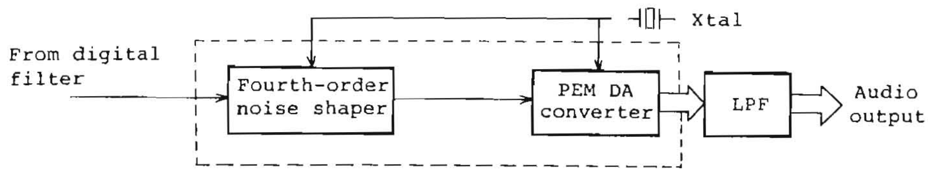


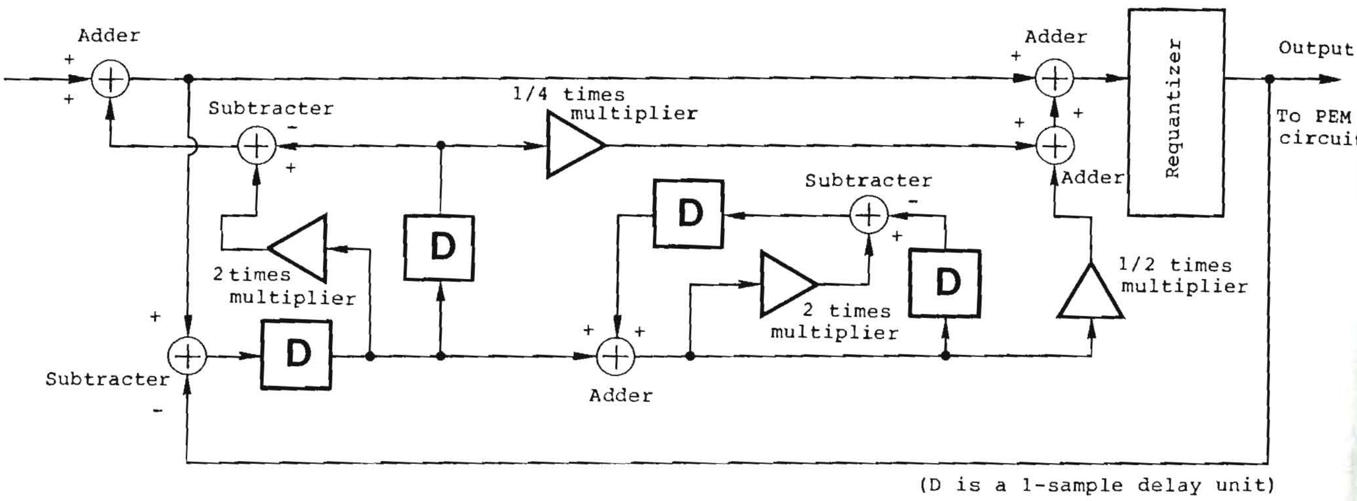
Fig. 3 Block diagram of a PEM DD converter

- Fourth-order noise shaper

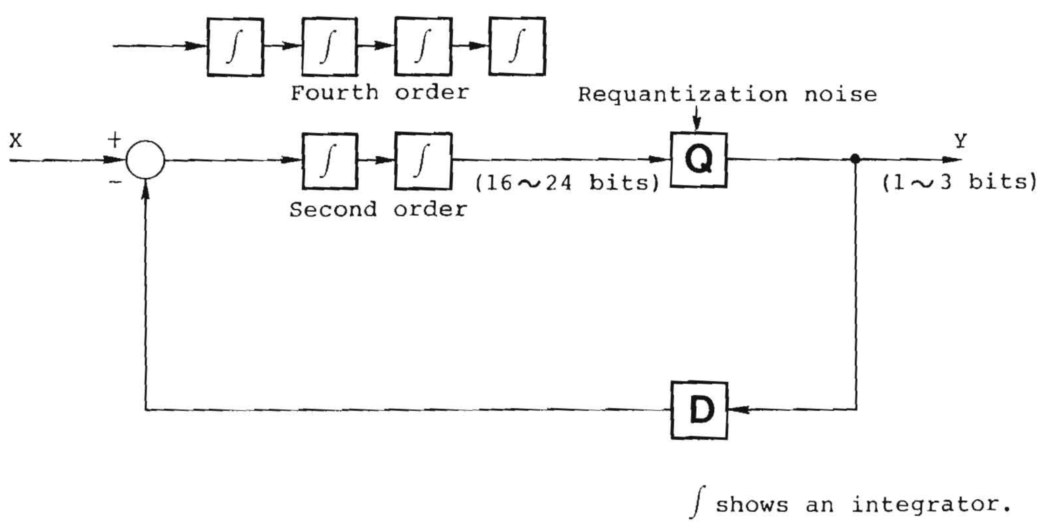
The circuit of a fourth-order noise shaper is shown in Fig. 4. In this circuit, data which is 8 times oversampled by a digital filter is input and it is further oversampled at a higher frequency. The noise shaper circuit operates at this higher sampling frequency (fa).

Q in Fig. 4 is a circuit called a "requantizer" which performs bit compression so that high-resolution data is compressed to data suitable for the 1-bit DA converters in the following stage. Distortion due to compression errors, called "requantization noise", always occurs after bit compression. The noise shaper's task is to remove this distortion so that the errors do not affect the output after DA conversion.

As shown in Fig. 4, the noise shaper consists of a negative feedback loop including the requantizer (Q). Requantization noise occurring in the loop is suppressed by the effects of negative feedback. The greater the loop gain, the greater the noise suppression effect. In the noise shaper, integrating



(a) Simplified block diagram of one JVC noise shaper



(b) Simplified block diagram of a noise shaper

Fig. 4 Fourth-order noise shaper

loop filters the gain of which increases at lower frequencies are used to suppress noise in the audible band, that is at lower frequencies. Therefore, the greater the degree of integration, that is the higher the order of the noise shaper, the greater its noise suppression effect.

In conventional 1-bit DA converters, when the order of noise shaping is increased, the requantizer is required to have a greater dynamic range. On the other hand, the local 1-bit DA converter in the following stage had low resolution, which limited the dynamic range. Because of this, the requantizer became saturated and the feedback loop became unstable.

However, even though the order of noise shaping is increased and the S/N in the audible range is improved, the noise shaper can only change the distribution of the noise. That is, it reduces noise at lower frequencies but noise at higher frequencies is increased. So, in JVC's noise shaper, a special loop filter is used so that at lower frequencies, sufficient negative feedback can be obtained to achieve the characteristics of fourth order noise shaping while at high frequencies, the dynamic range of the input to the requantizer is reduced and stable negative feedback can be obtained to achieve the characteristics of second order noise shaping. By use of this loop filter and our new high-resolution PEM DA

converters, an unsaturated, completely stable noise shaper with fourth-order characteristics can be achieved.

The characteristics of noise shaping are shown in Fig. 5. As you can see, noise in the audible range is suppressed with respect to the original characteristics of requantization noise as the order of noise shaping is increased. In this PEM DD converter, with its fourth-order noise shaper and PEM DA converters, requantization noise is reduced to a completely negligible level.

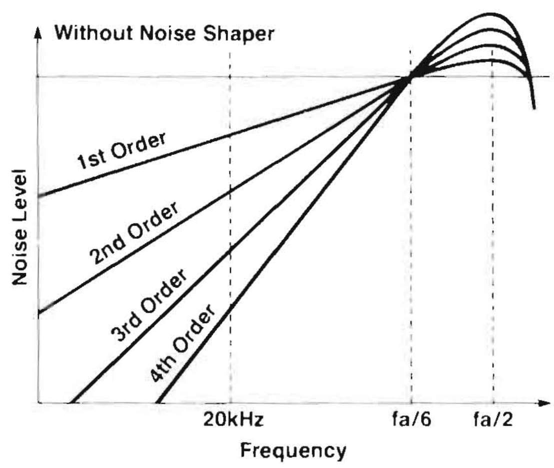


Fig. 5 Effect of a noise shaper

- PEM DA conversion system (Fig. 6)

The PEM DA conversion system converts input digital data into an extremely precise pulse train under the control of an accurate quartz clock, for conversion into an analog music signal by the subsequent LPF. The PEM DD converter contains two PEM DA converters which produce two pulse outputs (A and B), based on the input digital data, and these are synthesized to produce the output signal.

The basic operation of the PEM DA conversion system is as follows. If data +n is input, the edge moves by n clock periods in the direction of increasing pulse width in the case of DA converter A, while in the case of DA converter B, the edge moves by n clock periods in the direction of decreasing

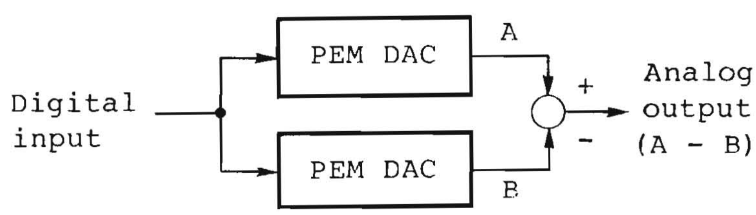


Fig. 6 Basic construction of a PEM DA converter

pulse width. When data -n is input, the opposite occurs. As the composite signal (A - B) is symmetrical on both sides of the reference timing (shown by the dotted line), and on both sides of the voltage axis (as can be seen from data +1 and -1), phase distortion does not occur.

In Fig. 7, when data +1 is input from the noise shaper, the edge of the pulse from DA converter A is shifted by one clock period before the reference timing shown by the dotted line, while the edge of the pulse from DA converter B is shifted by one clock period after the reference timing.

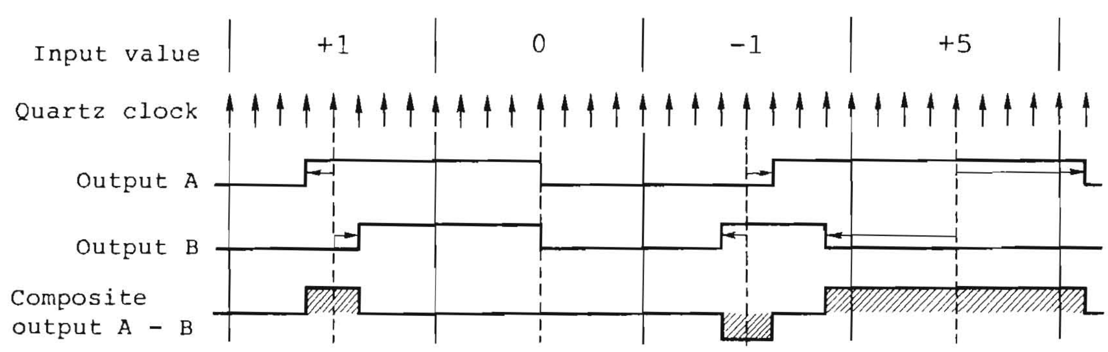


Fig. 7 Basic waveforms in a PEM DA converter

The difference from a conventional 1-bit DA converter is that, when the pulse stream is input, data is extracted using the pulse width in the case of a conventional DA converter, while in the case of the PEM DA converter, data is extracted using the edge position of the pulse. By extracting data in this way, using the edge position rather than the width of the pulses, the PEM DA converter has a resolution more than twice that of a conventional DA converter.

Also, with the conventional DA converter, the width can change only within the input data period (the area between full lines with the dotted line at the center), while, with the PEM DA converter, the only precondition is that the edges of the pulses from the two DA converters do not fall on each other.

As you can understand from the third data -1 and the fourth data +5 in the example shown in Fig. 7, edges can extend beyond the full line which is the border between adjoining data samples. For this reason, the PEM DA converter has a wider dynamic range.

5) <u>Conversion characteristics of the PEM DD converter (Fig. 8)</u>

The advantages of the PEM DD converter are:

1. Distortion which degrades sound quality such as zero-crossing distortion is completely eliminated.
2. Optimum, stable music reproduction is possible because it is not influenced by aging and temperature changes.
3. Low distortion (0.0015% with 16-bit input) thanks to improved linearity and quartz crystal accuracy
4. Requantization noise of -120 dB (logical value in audio band) and wide dynamic range with high resolution equivalent to 20 bits (0 - 20 kHz)

The photographs show the conversion characteristics at a very low level. The input signal consists of repetitions of square-wave signals, with each of the 5 series of square waves having an amplitude difference equal to the amplitude of the

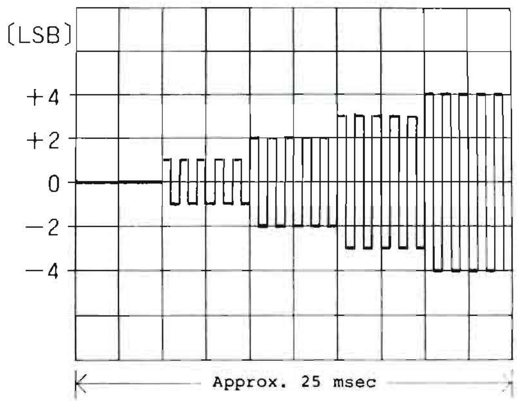
LSB. The conversion output passes through the same LPF and no averaging was applied for any of the three photographs.

Photo 1 shows the waveform when a current-summing DA converter is used and significant mismatching occurs between 0 and -1. This shows that zero-crossing distortion occurs and the linearity at low levels is degraded.

Photo 2 shows the waveform when a conventional 1-bit DA converter is used. The linearity is satisfactory but there is a significant amount of requantization noise. This decreases the dynamic range at the levels used for listening.

Photo 3 shows the conversion characteristics of JVC's PEM DD converter. This produces results which are closest to the input signal while noise is minimized and perfect linearity is maintained.

As described above, JVC's PEM DD converter is the ideal DA converter with accurate linearity even at extremely low levels, and a wide dynamic range.



(The value of the LSB is the smallest possible data increment)

Fig. 8 Input data (16 bits)

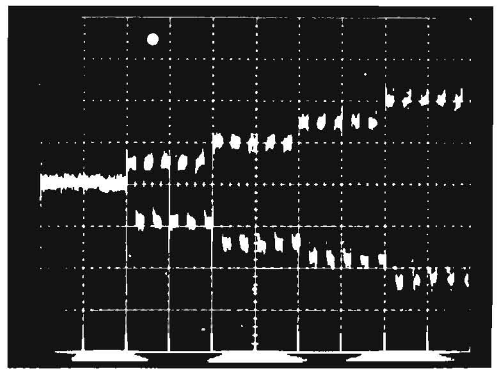


Photo 1 Conventional current-summing DA converter

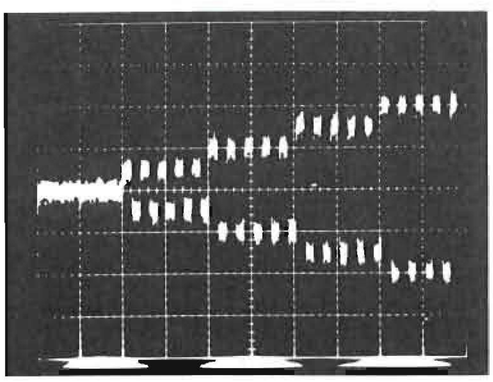


Photo 2 Conventional 1-bit DA converter

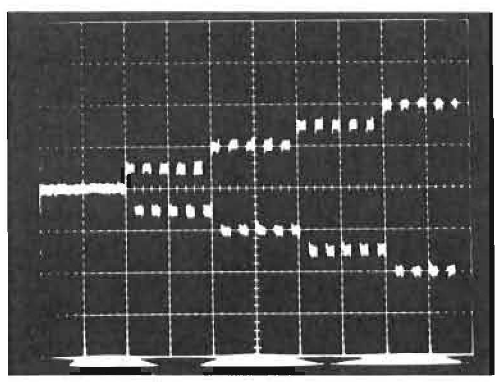


Photo 3 Conversion characteristics of the PEM DD converter

JVC

VICTOR COMPANY OF JAPAN, LIMITED
Audio Products Division

U.S. Version

Printed in Japan
SBU-0063